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AN INVESTIGATION OF DEVIATION CONTROL
FOR FREQUENCY MODULATION
COMMUNICATIONS SYSTEMS

JOHN THEODORE GEARY

1953

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AN INVESTIGATION OF DEVIATION CONTROL
FOR FREQUENCY MODULATION
COMMUNICATIONS SYSTEMS

by

John Theodore Geary,
Lieutenant, United States Navy

Submitted in partial fulfillment
of the requirements
for the degree of
MASTER OF SCIENCE
IN
ENGINEERING ELECTRONICS

United States Naval Postgraduate School
Monterey, California
1953

G257

This work is accepted as fulfilling
the thesis requirements for the degree of

MASTER OF SCIENCE

IN

ENGINEERING ELECTRONICS

from the

United States Naval Postgraduate School

PREFACE

The problem of interference from adjacent channel modulation in frequency modulation communications systems has become acute with the growth of the commercial radio services and the increasingly crowded spectrum. The investigation of deviation control is intended to give an understanding of this source of interference together with techniques for its decrease.

The project was conducted at the Phoenix Research Laboratory of Motorola, Inc., where the author was assigned by the U. S. Naval Postgraduate School as an integrated part of the Engineering Electronics curriculum.

The author wishes to express his gratitude to Motorola, Inc., for its co-operation in sponsoring the investigation, and in particular to Dr. Robert E. Samuelson for suggesting the topic and directing its initial course, to Mr. Russell M. Yost for his invaluable guidance and countless helpful suggestions, to the Messrs. Charles Estes, Douglas Reeves, Corwin Wilson, Paul Greer, Pete Nelson, and the many others for their friendly counsel and assistance.

PREFACE

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LIST OF ILLUSTRATIONS

Symbols and Abbreviations

mc	megacycles per second
v.h.f.	very high frequency
kc	kilocycles per second
F.C.C.	Federal Communications Commission
cps	cycles per second
db	decibel(s)
r.f.	radio frequency
i.f.	intermediate frequency
deviation	peak difference between the instantaneous frequency of the modulated wave and the carrier frequency
$e_c(t)$	instantaneous carrier voltage
E_c	maximum carrier voltage
E_m	maximum audio voltage
ω_c	angular frequency of carrier
ω_m	angular modulating frequency
k_f	constant of proportionality relating modulating voltage to frequency
$f(t)$	instantaneous frequency
f_c	carrier frequency
f_m	repetition frequency of modulation
f_D	deviation
$\phi(t)$	instantaneous phase

m_f	modulation index, frequency modulation
m_p	modulation index, phase modulation
m	modulation index, phase or frequency modulation
n	order of harmonic of modulating frequency
α	$\frac{m}{\pi}$
β	$n - m$
ν	$m + n$
ϵ	$-m$
sgn	algebraic sign
$C(z)$	Fresnel integral
$S(z)$	Fresnel integral
FM	frequency modulation
PM	phase modulation
IDC	Instantaneous Deviation Control

SUMMARY

The purpose of this paper is to investigate the problem of reducing sideband interference for frequency-modulated signals in the interest of maximum utilization of the r.f. spectrum by the commercial and government radio services.

The investigation is conducted by first developing a theoretical analysis of the radio frequency spectrum for frequency modulation by the following modulating waveforms, which are of interest when the speech is limited before modulating the transmitter: sinusoidal, complex, square wave, and triangular. It is shown that transmitted bandwidth depends on the deviation, repetition frequency, and the waveform of the modulating voltage.

Techniques are considered which will control the deviation, and, in accordance with the above analysis, will reduce sideband splatter. These methods are: the Instantaneous Deviation Control, a circuit developed and used commercially by Motorola; a double-integration technique; and an audio channel separation system. Measurements indicate that the channel separation method is somewhat superior in performance; the Instantaneous Deviation Control, when modified by a low-pass filter, is most desirable from the viewpoint of effective deviation limiting with simple circuitry.

CHAPTER I

INTRODUCTION

The Federal Communications Commission assigns frequencies in the 25-50 mc, 72-76 mc, and 152-174 mc bands in the v.h.f. spectrum for the use of commercial and governmental radio services. These services, which include mobile, land mobile, and fixed transmitting stations, are operated by public safety, land transportation, military, and industrial users. Channel widths of 40 kc are assigned in the two lower bands and 60 kc in the higher band. The number of transmitters licensed on these frequencies has multiplied many times since 1940, until now there are considerably more than 100,000 units in operation. In the past, unoccupied "guard" channels rendered the interference problem insignificant, but with the rapid growth of these services the problem has become extremely acute. Adjacent channels in the same area now must be utilized, and saturation of all channels in the near future is predicted.

Pertinent F.C.C. regulations[1] relating to commercial frequency modulation services are summarized as follows. Bandwidth is defined as the band of frequencies which contain 99% of the radiated power; 40 kc is the maximum authorized bandwidth. Spurious emission at a frequency separation from the carrier equal to the bandwidth shall in the case of low and medium-powered transmitters be attenuated at least 60 db. Transmission of modulation frequencies higher than 3000 cps is unauthorized. The maximum frequency deviation shall

CHAPTER I

INTRODUCTION

[illegible]

[illegible] F.C.C. regulations[1] [illegible] 2000 mc [illegible] not exceed 15 kc, and each transmitter must be provided with a device which limits the deviation to this value.

Intensive engineering research is being devoted to the problem of maximum utilization of the spectrum. Field tests have shown that factors of most concern in the reduction of adjacent or split channel interference are:

(a) on-frequency noise produced by the adjacent channel transmitter;

(b) desensitization of receivers produced by saturation of the adjacent channel carrier;

(c) interference caused by adjacent channel modulation;

(d) intermodulation interference, caused by the mixing of two strong signals and a resultant beat on the desired signal frequency.

From a study of the overall problem, an analysis by the Radio-Television Manufacturers Association[2] indicates that transmitter noise is the limiting factor, since the noise output from a typical phase-modulated transmitter is approximately 80 db below the unmodulated carrier for several hundred kc separation.

Recent engineering advances indicate that split channel operation, using 20 kc channel spacing with a maximum deviation of 5 kc, is feasible. Research by several leading laboratories predicts eventual improvement of the transmitter noise emission to -100 db. The problems of desensitization and intermodulation interference

are, in general, being solved in the receiver by the use of low gain r.f. stages and highly selective filters preceding high gain i.f. stages. Reduction of undesired sideband emission, then, becomes of major, if not primary importance in adjacent channel or future split channel operation.

CHAPTER II

THEORETICAL ANALYSIS

1. Mechanism of Interference

It is important to consider first the cause and mechanism of interference from adjacent channel modulation.[3]

Sidebands from the adjacent channel transmitter on the desired channel cause interference as a result of audible beat frequencies between the desired and undesired signals. The beat frequency frequency-modulates the desired signal during the process of limiting in the receiver. This interference increases with the ratio of the peak amplitude of the interfering sideband to the peak of the desired signal and, in addition, with an increase in the beat frequency.

When the beat frequency is beyond the audio range, interference occurs when the instantaneous frequency from the interfering station extends into the nonlinear part of the receiver discriminator curve. Because of the steep sides of the receiver selectivity curve, amplitude variation of the interfering signal results, thereby producing audible interference.

Thus interference from adjacent channel modulation is a function of both the bandwidth of significant sideband amplitudes and the maximum frequency excursion of the interfering transmitter.

2. General

In order to understand the correlated problems of deviation

and bandwidth, it is necessary to examine the theoretical spectrum produced by frequency-modulating an r.f. carrier.

In frequency modulation, the frequency of a carrier is varied by an amount that is proportional to the amplitude of the modulating signal and at a rate that is proportional to the frequency of the modulating source. The maximum swing of the modulated carrier frequency from its mean value is called the "frequency deviation," or more briefly, the "deviation." Frequency modulation of a radio frequency carrier produces an infinite number of sidebands which are spaced from the carrier and from each other by the frequency of the modulating source in the case of single frequency sinusoidal modulation, and by all possible sum and difference frequencies in the case of complex modulation.

The general expression for an unmodulated carrier is

$$e_c(t) = E_c \sin(\omega_c t + \theta) \tag{1}$$

The quantity $\omega_c t + \theta$ is the instantaneous phase of the carrier, $\phi(t)$. Thus,

$$\phi(t) = \omega_c t + \theta, \text{ and} \tag{2}$$

$$e_c(t) = E_c \sin \phi(t) \tag{3}$$

It is important to note that the angular frequency is the time derivative of the phase and, accordingly, may be expressed by

$$\omega(t) = \frac{d}{dt} \phi(t). \tag{4}$$

3. Single-Frequency Sine Wave

(a) Frequency modulation

Considering first frequency modulation by a single-frequency sine wave, the instantaneous angular frequency is, by definition,

$$\omega(t) = \omega_c + k_f E_m \cos\omega_m t \tag{5}$$

where k_f is a constant relating voltage to frequency, and $E_m \cos\omega_m t$ is the modulating signal. The instantaneous frequency is, therefore,

$$f(t) = f_c + \frac{k_f E_m}{2\pi} \cos\omega_m t \tag{6}$$

and the maximum excursion, or frequency deviation, is

$$f_D = \frac{k_f E_m}{2\pi} \tag{7}$$

Therefore in the case of a frequency-modulated signal the deviation is proportional to the amplitude of the modulating voltage and is independent of modulating frequency.

Since $\phi(t) = \int_0^t \omega(t)dt$, the instantaneous phase is

$$\phi(t) = \omega_c t + \frac{k_f E_m}{\omega_m} \sin\omega_m t \tag{8}$$

Defining the modulation index for frequency modulation, m_f, by the relation

$$m_f = \frac{k_f E_m}{\omega_m} \tag{9}$$

we have

$$e_c(t) = E_c \sin(\omega_c t + m_f \sin\omega_m t) \tag{10}$$

The modulation index is related to the deviation and modulating frequency by

$$m_f = \frac{f_D}{f_m} \tag{11}$$

where f_m is the modulating frequency.

(b) Phase modulation

Phase modulation is produced by varying the instantaneous phase of the carrier at a rate proportional to the modulating frequency and by an amount proportional to the amplitude of the modulating voltage. Hence, by definition,

$$e_c(t) = E_c \sin(\omega_c t + k_f E_m \sin\omega_m t) \tag{12}$$

where the modulating voltage is $E_m \sin\omega_m t$. By defining m_p as the modulation index for phase modulation,

$$e_c(t) = E_c \sin(\omega_c t + m_p \sin\omega_m t). \tag{13}$$

It therefore follows that the instantaneous frequency is

$$f(t) = f_c + m_p f_m \cos\omega_m t \tag{14}$$

and the deviation is

$$f_D = m_p f_m \tag{15}$$

In the case of a phase-modulated wave, the deviation is proportional to both the amplitude and the frequency of the modulating voltage, and the modulation index is proportional to the amplitude of the modulating voltage.

It can be seen that, with the differences noted above, frequency modulation and phase modulation are indeed similar. In either case the carrier frequency is varied at a rate proportional to the modulating frequency. It is significant that the sideband amplitudes of a frequency-modulated spectrum are identical to those of a phase-modulated spectrum that is produced by differentiating the modulating voltage before phase modulating.

In the following discussion no distinction will be made between frequency modulation and phase modulation. The actual difference is in the interpretation of the modulation index, which for the general case will be symbolized by m.

The expression $e_c(t) = E_c \sin(\omega_c t + m \sin\omega_m t)$ may be expanded in a spectrum consisting of a carrier and sidebands and written as follows[4]:

$$\begin{aligned} e_c(t) = {} & J_0(m)\, E_c \sin\omega_c t \\ & J_1(m) E_c \left[\sin(\omega_c + \omega_m)t - \sin(\omega_c - \omega_m)t\right] \\ & J_2(m) E_c \left[\sin(\omega_c + 2\omega_m)t + \sin(\omega_c - 2\omega_m)t\right] \\ & J_3(m) E_c \left[\sin(\omega_c + 3\omega_m)t - \sin(\omega_c - 3\omega_m)t\right] \\ & + \ldots\ldots \end{aligned} \tag{16}$$

This may be more concisely expressed as

$$e_c(t) = E_c \sum_{n=-\infty}^{\infty} J_n(m)\sin(\omega_c t + n\omega_m t) \tag{17}$$

where $J_n(m)$ is a Bessel function of the first kind of order n and argument m. It is obvious that the sideband separation is equal to the modulating frequency.

The sideband amplitudes may be plotted with the use of a table of Bessel functions. The number of significant sidebands increases with the modulation index. Because the total energy in the spectrum is not changed by modulation, the average amplitude of the significant sideband components decreases with spectra of increasing modulation index. Although there is an infinite number of side frequencies in the spectrum, the amplitudes diminish rapidly at frequency separations greater than the deviation.

Figures (1) and (2) show the correlation between deviation and bandwidth for sine wave modulation.* From the system of curves in Figure (1) the envelope of sideband amplitudes may be plotted for modulation indices between 0.1 and 100. It is readily apparent that significant sideband amplitudes extend well beyond the deviation. For a constant modulating frequency, increasing the deviation increases the half-bandwidth, which approaches the deviation for large excursions. For example, a sine wave of frequency 500 cps and deviation of 5 kc has a -60 db half-bandwidth of 7 kc; increasing the deviation, say,

*Figures 1 through 5 are plotted by using the curves and equations given by Corrington[4].

that may be [illegible] represented as :

$$e_o(t) = E_o \sum_{n=-\infty}^{\infty} J_n(\beta)\cos(\omega_c + n\omega_m)t \qquad (19)$$

where $J_n(\beta)$ is Bessel function of the first kind of order n and argument β. It is implied that the sideband separation is equal to the modulating frequency.

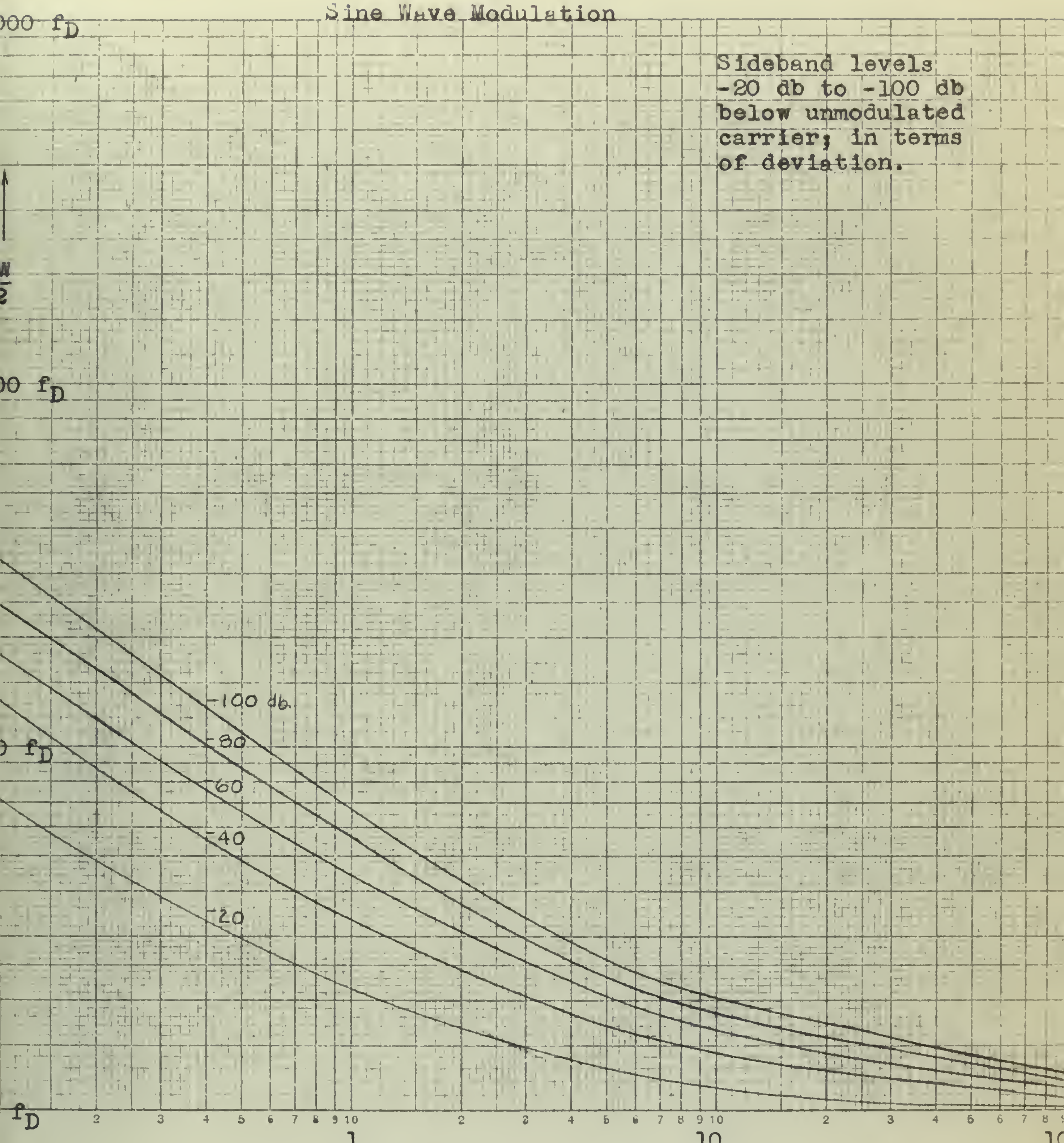
The sideband amplitudes may be [illegible] with the aid of a table of Bessel functions. The number of significant sidebands increases with the modulation index. Because the total energy in the spectrum is not changed by modulation, the average amplitude of the significant sideband components decreases with [illegible] of increasing modulation index. Although there is an infinite number of side frequencies in the spectrum, the significant sideband [illegible] is [illegible] than the bandwidth.

Figures (1) and (2) show the [illegible] between [illegible] and [illegible] for high and [illegible]. From the system of curves in Figure (1) the spectrum of [illegible] modulation may be plotted for modulation indexes between 0.5 and 100. It is readily apparent that significant sideband [illegible] extend well beyond the deviation. For a constant modulating frequency, increasing the deviation increases the bandwidth, which approaches the deviation for large deviations. For example, a [illegible] of [illegible] and deviation of [illegible] has a [illegible] bandwidth of [illegible].

*Figures 1 through 3 are obtained by using the curves and equations given by [illegible].

Half-Bandwidth
vs.
Modulation Index

Sine Wave Modulation



$$m = \frac{f_D}{f_m}$$

Fig. 1

Half-Bandwidth
vs.
Deviation

Sine Wave Modulation

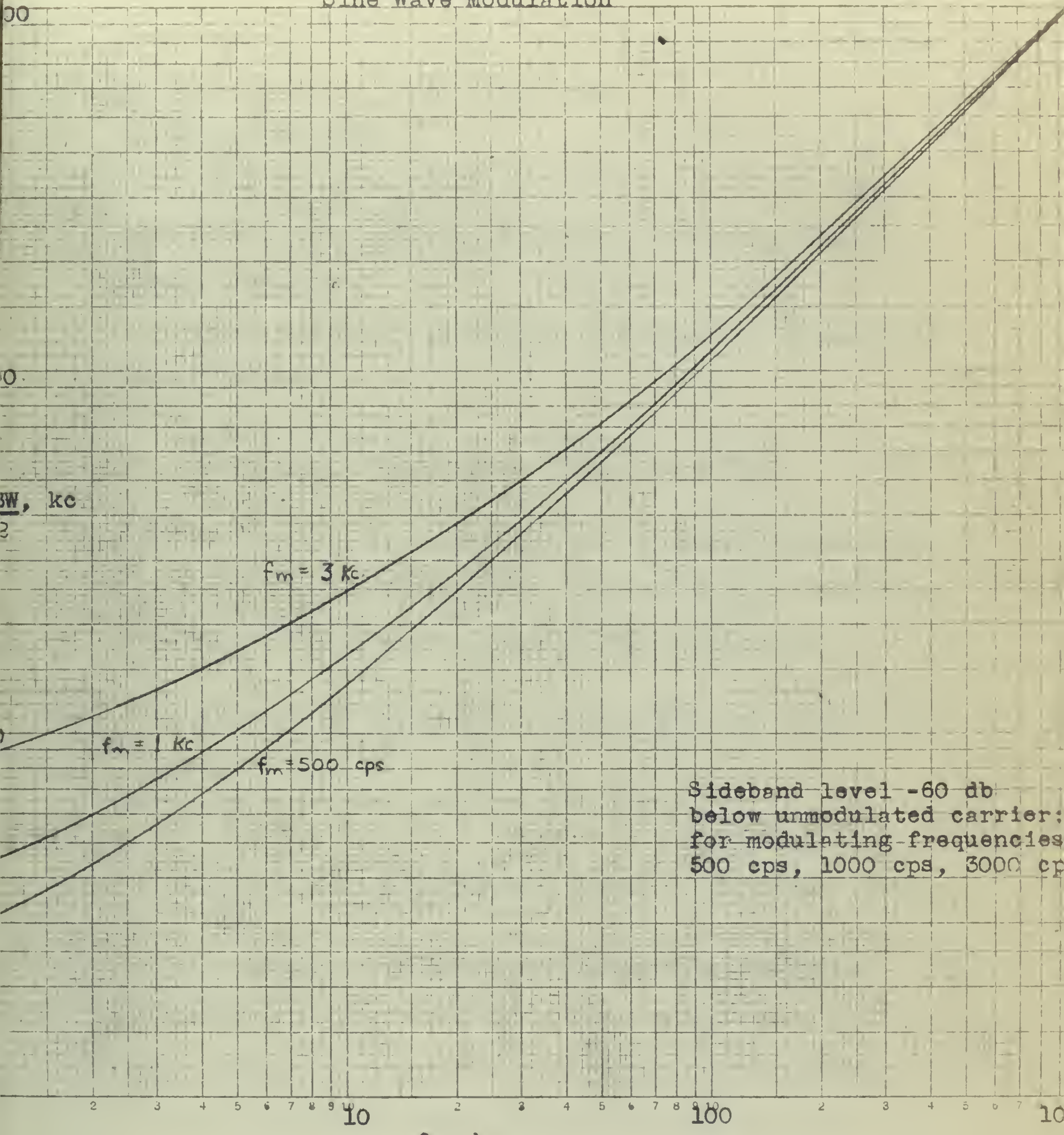


Fig. 2

5 times increases the half-bandwidth to 31 kc. Figure (2) shows that increasing the modulating frequency while maintaining constant deviation results in increased bandwidth.

4. Complex Modulation

Because a speech waveform contains many discrete frequency components, it is of practical interest to consider modulation by a complex waveform.

If more than one audio frequency is present, the resulting carrier is given by

$$e_c(t) = E_c \sin\left[\omega_c t + \sum_{s=1}^{S} m_s \sin(\omega_s t + \epsilon_s)\right] \qquad (18)$$

where ω_s represents the modulating frequency, and ϵ_s the phase angle corresponding to ω_s. The spectrum, which is much more complicated than for a single modulating frequency, may be shown to be[4]

$$e_c(t) = E_c \sum_{k_s=-\infty}^{\infty} \left\{\prod_{s=1}^{S} J_{k_s}(m_s)\right\} \sin\left[\omega_c t + \sum_{s=1}^{S} k_s(\omega_s t + \epsilon_s)\right] \qquad (19)$$

In the case of two-tone modulation this becomes

$$e_c(t) = E_c \sum_{j=-\infty}^{\infty} \sum_{k=-\infty}^{\infty} J_j(m_1) J_k(m_2) \sin(\omega_c t + j\omega_1 + k\omega_2)t \qquad (20)$$

It is seen that sidebands will be produced at frequencies differing from the carrier by all possible combinations of $- jf_1 - kf_2$; or, in other words, by all combination frequencies that can be obtained by taking sum and differences of all the

harmonics of the modulating tones. The amplitude of each sideband is proportional to the product of the two Bessel functions having orders j and k equal to the orders j and k of the modulating frequencies. The significant bandwidth is approximately the sum of the two bandwidths which would be produced if each frequency were modulated separately. It is clear that the sidebands spaced symmetrically about the carrier are of equal amplitudes. Those sidebands which contain odd multiples of either modulating frequency have opposite phases on each side of the carrier.

5. Square Wave Modulation

In modulating a carrier by a speech waveform, excessive audio voltage will cause amplitude limiting in the pre-modulation stages of the transmitter. This will be true, of course, if intentional clipping or compression is used to control the deviation. In the limit, as the modulation is increased, clipping produces a carrier frequency-modulated essentially by a square wave (which is analogous to phase-modulating with a triangular wave). Hence, the square wave FM spectrum becomes of practical interest.

Corrington[4] describes the general method for computing sideband amplitudes for a given modulating signal in terms of its Fourier coefficients, and derives the general relation for rectangular frequency modulation. In particular, the sidebands for a square wave are given by

$$e_c(t) = \sum_{n=-\infty}^{\infty} \frac{2mE_c}{(m^2 - n^2)\pi} \sin\left[(m-n)\frac{\pi}{2}\right] \cdot \sin(\omega_c t + n\omega_m t) \qquad (21)$$

where $m \neq \pm n$. For $m = \pm n$,

$$e_c(t) = \tfrac{1}{2} \sin(\omega_c t + n\omega_m t) \tag{22}$$

It is interesting to note that the maximum sidebands occur where $m = \pm n$. For integral values of m, every other sideband is of zero amplitude, except for $m = \pm n$. The system of curves in Figure (3) shows that significant sideband amplitudes exist well beyond the deviation. For example, for m = 5, the sideband -60 db below the unmodulated carrier is eleven times the deviation.

6. Triangular Modulation

If a square wave is integrated, the resulting waveform is a triangular wave. A clipped or compressed speech wave may be integrated to produce triangular frequency modulation; hence, the spectrum for a triangular signal becomes of practical interest.

The sideband amplitudes are given by[4]

$$e_c(t) = \sum_{n=-\infty}^{\infty} \left\{ \frac{1}{\sqrt{2\pi\alpha}} \cdot \cos \frac{\beta^2}{4\alpha} \cdot \left[(\operatorname{sgn} \nu) C\left(\frac{\nu^2}{4\alpha}\right) - (\operatorname{sgn} \beta) C\left(\frac{\beta^2}{4\alpha}\right) \right] \right.$$

$$+ \frac{1}{\sqrt{2\pi\alpha}} \sin \frac{\beta^2}{4\alpha} \cdot \left[(\operatorname{sgn} \nu) S\left(\frac{\nu^2}{4\alpha}\right) - (\operatorname{sgn} \beta) S\left(\frac{\beta^2}{4\alpha}\right) \right]$$

$$\left. - \frac{1}{\pi\nu} \sin(\pi\nu + \epsilon) \right\} \cdot \sin(\omega_c t + n\omega_m t).$$

where $\alpha = \frac{m}{\pi}$, $\nu = m + n$, $\beta = n - m$, $\epsilon = -m$, $(\operatorname{sgn}\beta)$ and $(\operatorname{sgn}\nu)$ indicate the algebraic sign of β and ν respectively. C and S are the Fresnel integrals defined by

Half-Bandwidth
vs.
Modulation Index

Square Wave Modulation

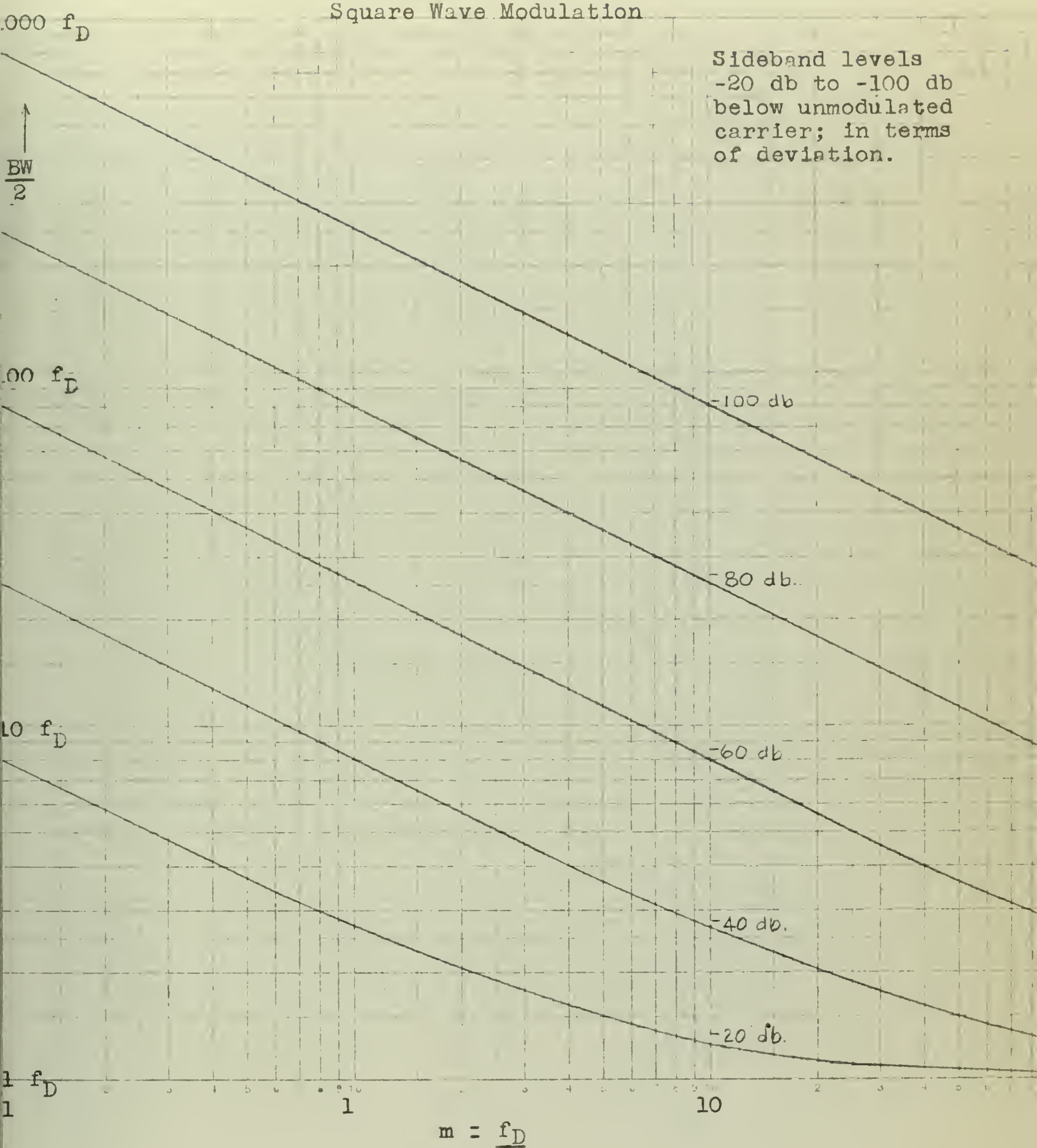


Fig. 3

Half-Bandwidth
vs.
Modulation Index

Triangular Modulation

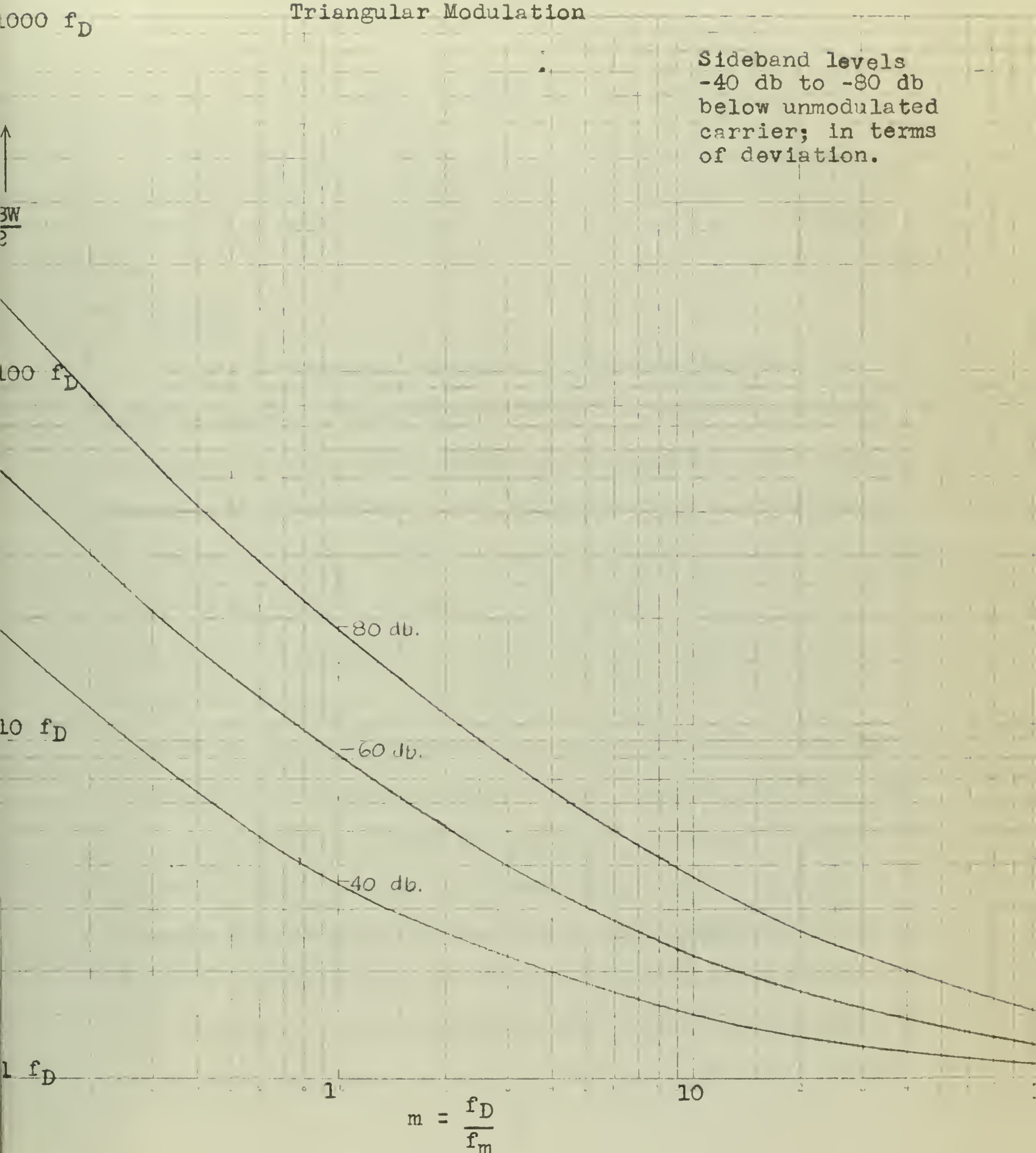


Fig. 4

$$C(z) = \frac{1}{\sqrt{2\pi}} \int_0^z \frac{\cos t}{\sqrt{t}}\, dt$$

$$S(z) = \frac{1}{\sqrt{2\pi}} \int_0^z \frac{\sin t}{\sqrt{t}}\, dt$$

Figure (4) shows a system of curves for various sideband levels produced by frequency modulating an r.f. carrier with a triangular wave.

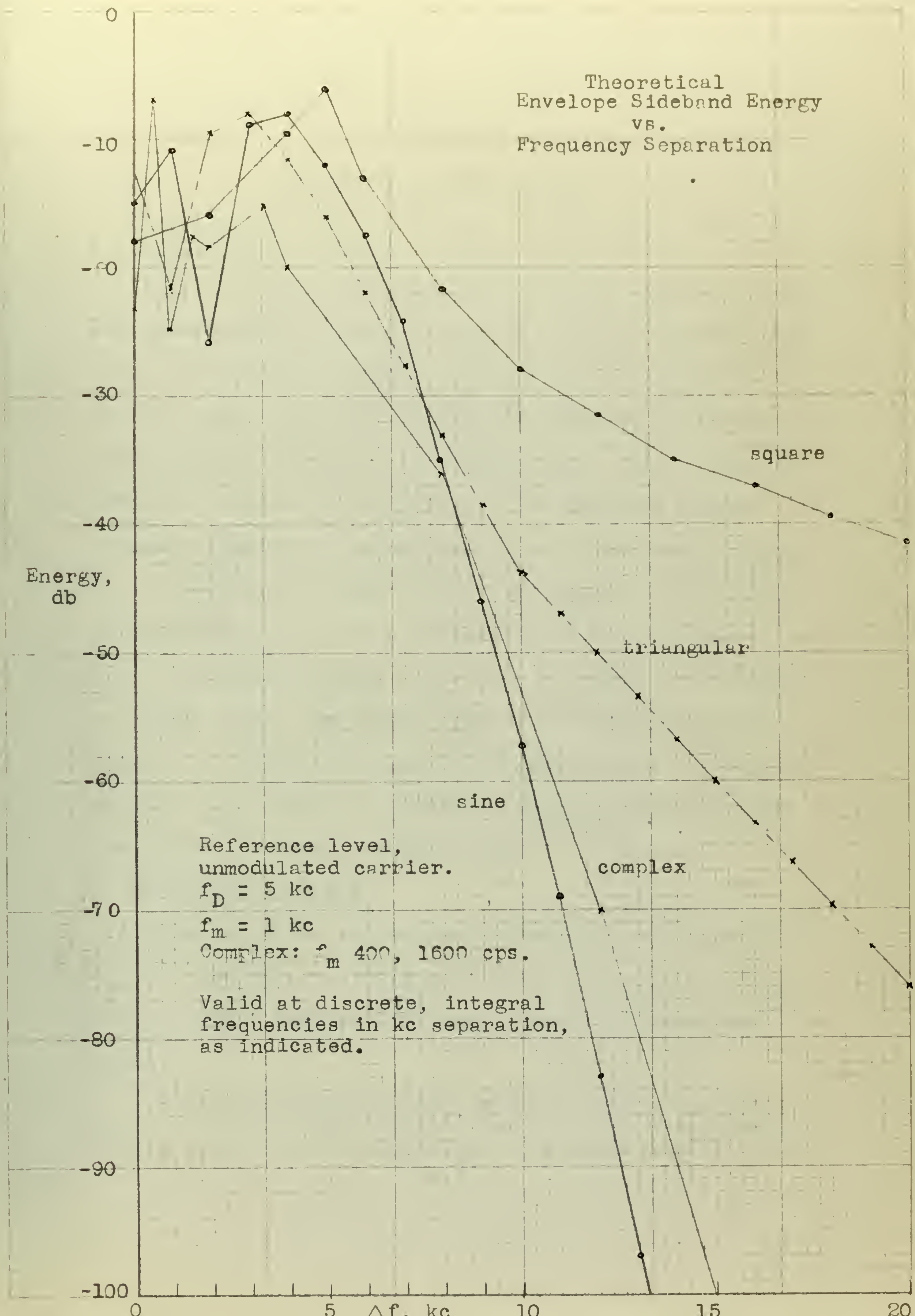
7. Summary

As can be seen upon examination of the above equations, an infinite number of sidebands exist whenever a signal is frequency-modulated. Sideband energy beyond the deviation diminishes rapidly, depending on the waveform of the modulating signal. By comparison of Figures (1), (3), and (4), it can be seen that for equal modulation indices the bandwidth in terms of the deviation is less for a sine wave than for a triangular wave and less for triangular than for square wave modulation. Considering, for example, a modulation index of 10, the half-bandwidth for a square wave is 10 times the deviation; for a triangular wave it is 2.3 times the deviation; and for a sine wave is 1.6 times the deviation. It follows from this comparison that the addition of harmonically related components increases the bandwidth for waveforms of equal deviation. It is therefore postulated that the relative amplitudes of the higher order harmonics of the modulating waveform, which determine both its maximum slope and abrupt corners, are responsible for the increased

bandwidth. This concept is important for understanding the effect of frequency-modulating with a complex waveform; if the higher frequency components are of such magnitude that the maximum slope of the waveform is greater than for a given sine wave with equal deviation, the complex waveform will conceivably have a greater bandwidth. This contradicts the explanation offered by Terman[5] that complex modulation tends to compress the bandwidth.

From the foregoing, we may conclude that the bandwidth of a frequency-modulated carrier is a function of (a), deviation; (b), repetition frequency of the modulating voltage; (c), wave-shape of the modulating voltage.

Of partical interest for 20 kc channel widths is a deviation of 5 kc. Figure (5) shows a plot of sideband energy for modulating waves of repetition rate 1000 cps, deviation 5 kc, and modulation index 5; complex modulation is shown by a waveform of deviation 5 kc for frequencies of 400 cps and 1600 cps in the voltage ratio of 2½:1. At 10 kc separation from the carrier, which corresponds to the edge of the adjacent channel, the sideband level for a square wave is -28 db, for a triangular wave -36 db, complex signal -52 db, and sine wave -57 db. At 20 kc separation, the adjacent channel center, the sideband levels for square and triangular waveforms are respectively -42 db and -76 db below the unmodulated carrier.



Theoretical
Envelope Sideband Energy
vs.
Frequency Separation
Energy, db
0
-10
-20
-30
-40
-50
-60
-70
-80
-90
-100
0
5
Δf, kc
10
15
20
square
triangular
sine
complex
Reference level, unmodulated carrier.
f_D = 5 kc
f_m = 1 kc
Complex: f_m 400, 1600 cps.
Valid at discrete, integral frequencies in kc separation, as indicated.

CHAPTER III

METHODS FOR LIMITING DEVIATION

1. Introduction

From the theoretical development it is clear that the deviation must be controlled or limited in order to restrict the emitted bandwidth. An additional important reason for deviation control is to keep the transmitted deviation within the pass band of the receiver, thus increasing the receiver signal-noise ratio. Since it has been shown that deviation is proportional to the modulating voltage, deviation control may be accomplished in the pre-modulation stage of the transmitter by proper design of the gain characteristics of the speech amplifier. Severe "overmodulation" may be caused, however, by an operator holding a microphone too closely or speaking too loudly; further, the dynamic range of the voice is such that variations of 10-20 db may occur in a single syllable. It is seen, therefore, that a method of amplitude limiting or compression must be employed in the speech amplifier stage.

2. Speech Characteristics

Before considering various practical means for amplitude limiting, it is desirable to discuss briefly the characteristics of the modulating speech waveforms upon which a practical circuit must operate.

For the reproduction of speech, "perfect" fidelity requires a frequency range of about 100-8000 cps, and a volume range of 40 db[6].

For communications systems, intelligibility, rather than perfect reproduction, is the criterion, and a much lower frequency range proves desirable from the viewpoints of both noise and transmitted bandwidth reduction. From data compiled by Olson, a syllable articulation of 86% is realizable with a low-pass filter of 3000 cps, and 98% with a high pass filter at 300 cps. This combined articulation corresponds to approximately 98% sentence intelligibility. Based on these considerations, the F.C.C. authorizes transmissions of only those audio frequencies below 3000 cps for the subject communications services.

Very large fluctuations of instantaneous level are present in speech. Peak-to-rms sound pressures vary from 10 db at 300 cps to 17 db at 3000 cps for an approximate syllabic period of 1/8 section,[7] indicating that sharp, low-energy peaks are characteristic of the higher frequency components. As an indication of the relative peak amplitudes in the spectrum of a typical male voice, the peak pressures at 300 cps are approximately 10 db greater than at 3000 cps.

Although as much intelligence is contained in the spectrum below 1300 cps as above, it is seen that much less energy is contained in the higher frequency components. The major energy content is in the vowel sounds, which lend character or quality to speech. The inflections which separate and render distinguishable different words with the same basic vowel sounds are produced by the consonants.

The effect of listener reaction to nonlinear distortion can be summarized by noting that for a 300-3000 cps communications system,[6] 1.5% distortion is perceptible, 10% is considered tolerable, and 16% objectionable.

3. Speech Clipping

Amplitude limiting may be accomplished by the rather severe means of clipping the peaks of the speech waveform at a preset level. This results in effective deviation control and may be accomplished with few components and simple circuitry.

From the preceding discussion, an important disadvantage is evident by the production, in the limiting case, of square-wave modulation. Referring to Figure (3), for example, a 500 cps square wave with deviation of 5 kc has a -60 db half-bandwidth of 40 kc, as compared to the corresponding sine wave, which half-bandwidth is 7 kc. In practice, however, a clipped speech wave will not be of rectangular shape, but will have a finite slope and a bandwidth somewhat less than that indicated for a square wave.

Clipping of a complex waveform tends to emphasize the voltages of greater amplitude and suppress the components of lesser magnitude. Without some speech pre-emphasis, undesirable suppression of the high frequency components, which have relatively low peak amplitudes and high articulation value, will result, thus reducing the intelligibility. However, with high-frequency emphasis previous to clipping, it is conceivable that articulation will not be significantly impaired

by the clipping process. Poor sound quality results from clipping because of the generation of harmonic distortion; hence, to preserve the quality of reproduction only a limited amount of clipping is permissable. It is important to note here that whenever amplitude limiting takes place, rigorous analysis of a system cannot be made on the basis of a single-frequency sine wave.

Amplitude limiting may be followed by a 3000 cps low-pass filter, which will remove the higher order frequency components produced by clipping and decrease the maximum slope of the modulating waveform. Thus undesired sideband emission will be reduced and harmonic distortion will be decreased. Harmonics of the lower modulating frequencies, however, will not be effectively attenuated by a filter designed to pass the entire audio spectrum below 3000 cps; hence, harmonic distortion, though reduced, will still be a limiting factor.

4. Speech Compression

It is of interest to consider the operation of a compressor circuit, which might conceivably be used to limit the peak amplitude of the modulating voltage. A conventional compressor acts rapidly on a peak signal to reduce the circuit amplification so as to maintain a constant peak output level for an increase in input above a given level. Once the gain has been reduced it must return to normal at a relatively slow rate, otherwise severe distortion will be introduced on the wave immediately following the compressed peak.

In standard broadcast compressors, 90% recovery does not occur for 2 to 7 seconds. This interval may be reduced for speech but should not be so rapid that it is equal to the syllabic rate. As the majority of syllables in speech contains at least one peak it may be seen that the gain will be reduced for a major portion of time. Also, the finite operating time required by the limiter will allow initial peaks to pass unattenuated and thus cause undesired sideband interference. Further, compressing, or reducing the amplitude of the peaks is analogous to clipping as previously described. A further disadvantage is in the circuit complexity for a suitable compressor.

An instantaneous compressor operating on the principle of a logarithmic amplifier is feasible, but from the standpoint of deviation limiting would not prove as satisfactory as a conventional clipper. The maximum slope of the modulating waveform might conceivably be decreased from that obtained by clipping, but that advantage would be more than offset by the permitted variation in deviation.

5. <u>Instantaneous Deviation Control</u>

A practical speech clipping technique is illustrated by the IDC, which is used commercially by Motorola in its mobile transmitters. The circuit, developed by M. R. Winkler,[8] is shown in Figure (6).

The IDC consists of a 6 db per octave pre-emphasis, or differentiating network, followed by a clipper and a -6 db per octave de-emphasis, or integrating network. For modulating voltages below the clipping level, the differentiator-integrator reproduces the

Block Diagram

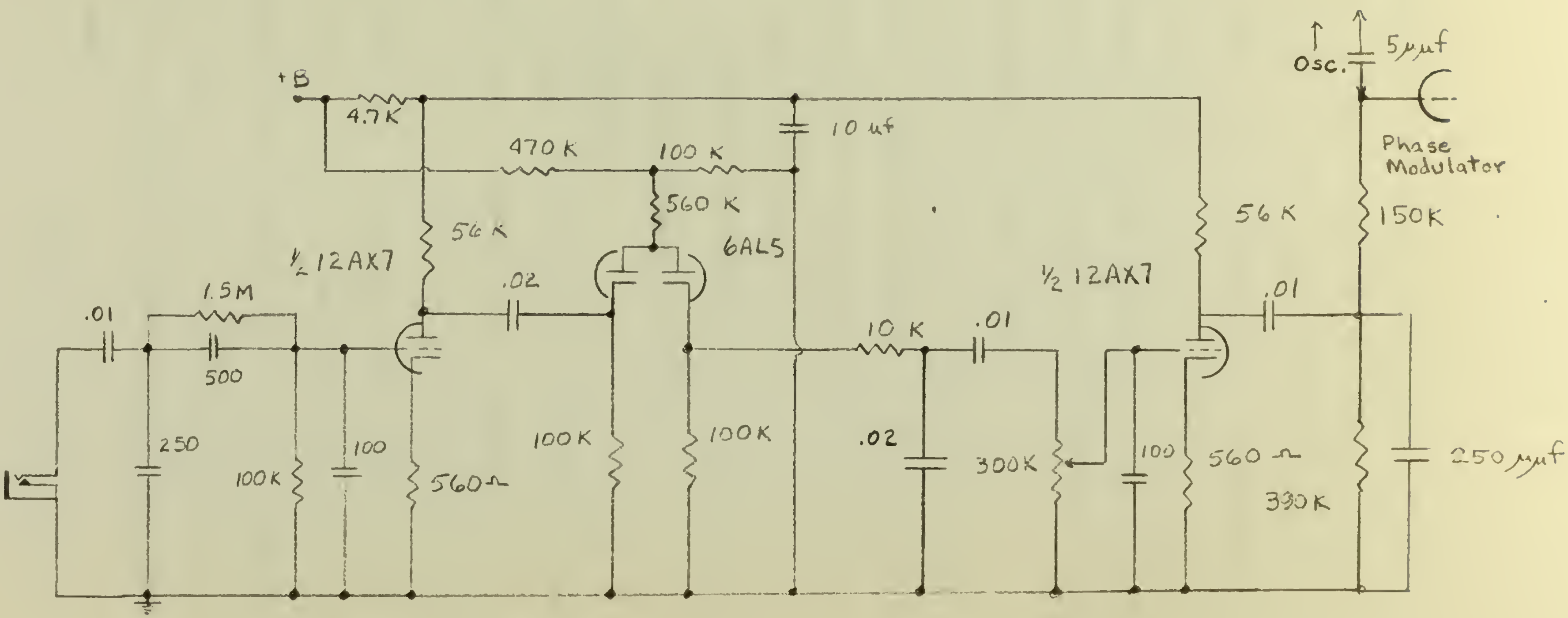


Fig. 6 Instantaneous Deviation Control

natural speech emphasis. For modulating voltages greater than the preset clipping level, the pre-emphasis raises the level of the high-frequency components so that suppression effects due to clipping will be minimized. In the limiting case of a square wave, the integrating circuit produces a triangular waveform. Besides the theoretical improvements in sideband emission due to deviation control, harmonic distortion is considerably reduced by the integration, with resulting improvement in sound quality.

In practice, however, this system is applied to a phase-modulated transmitter. Although effective in limiting the deviation, the IDC theoretically results in increased sideband splatter. It is analogous to frequency modulating clipped speech as described in paragraph (II-3). Sideband interference may feasibly be reduced with the addition of a 3000 cps low-pass filter.

6. Double Integration Method

As previously stated, the transmitted bandwidth for a frequency-modulated triangular wave is significantly less than for a square wave. It follows from the discussion of the IDC that a means might be devised for synthetically producing a wave which, when applied to a phase modulator, will produce a triangular frequency deviation. This can be accomplished by the addition of a second integrating stage following the clipper circuit, and necessitates a second differentiation preceding the clipper in order to reproduce a flat response for an unclipped signal.

From considering the effect of waveshape on bandwidth reduction, this technique is desirable. Further examination indicates, however, that deviation limiting will not be as effective as for the IDC. This is shown by noting that the integral of two square waves of equal amplitude but different repetition frequencies is greater for the signal of lower frequency (longer period). As an example, compare equal amplitude square waves of frequencies 400 cps and 1200 cps. The area under the former is three times that of the latter; hence, the deviation will be three times as great. Upon examination of Figures (3) and (4), it is seen for this example that despite the increased deviation, modulating with a triangular waveform decreases the transmitted bandwidth.

The pre-emphasis circuit theoretically emphasizes the higher frequency components previous to clipping such as to suppress the low frequencies. An exact analysis is difficult, and it remains to develop and test a circuit embodying these principles.

7. <u>Channel Separation</u>

Another means of speech clipping is conceivable by separating the modulating spectrum into bands or octaves, clipping and filtering each channel separately, and recombining. For example, the speech may be filtered to produce 200-400 cps, 400-800 cps, 800-1600 cps, and 1600-3000 cps channels. Use of sharp low-pass filters following the clipping stages will produce, upon recombination of the several channels, essentially very little harmonic distortion of the original

modulating waveform. Obviously, the greater the number of channels the more exactly will the speech be reproduced. Theoretically, this method appears to be relatively a high-quality system of low distortion, good deviation control, and excellent bandwidth characteristics. It suffers from undesirable circuit complexity. Reducing the number of channels reduces the complexity but in turn reduces the effectiveness of the method.

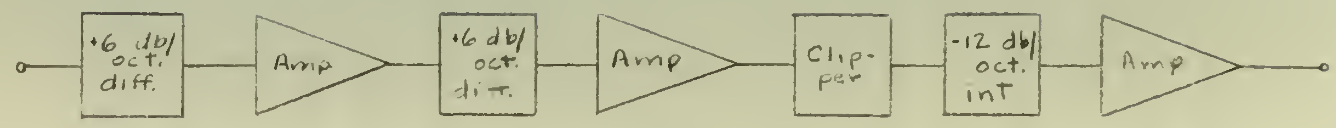
8. Circuit Design

Design features of laboratory models of speech clipping circuits embodying the principles of the double integration and channel separation methods are discussed in the following paragraphs.

(a) Double Integrator Circuit

As seen from the block diagram, Figure (7), the input network and first amplifier stage are cascaded differentiating circuits, each of time constant 50 microseconds, which provide an increasing response of 12 db per octave to 3200 cps. The circuit diagram, Figure (7),* shows a series clipper that utilizes vacuum instead of germanium diodes to increase the range of operation without "break-through", arranged in a novel manner to eliminate coupling capacitor and deleterious charging effects upon clipping. The clippers are followed by an integrating network of time constant 1000 microseconds that provides 12 db per octave attenuation from 150 cps. The 100 K potentiometer controls the symmetry of clipping;

*This is a slight modification of the circuit tested. It will provide integration with steeper roll-off; load impedance for the 6AU6 is increased, thus providing additional gain.



Block Diagram

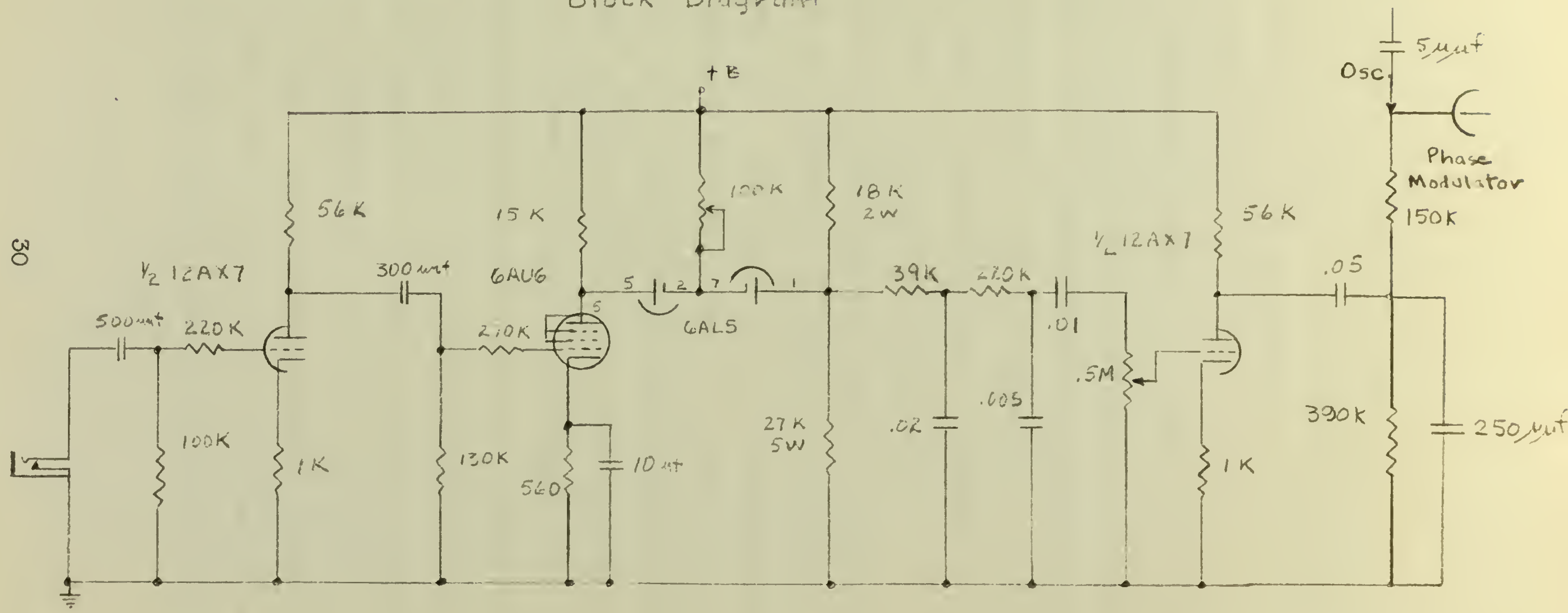


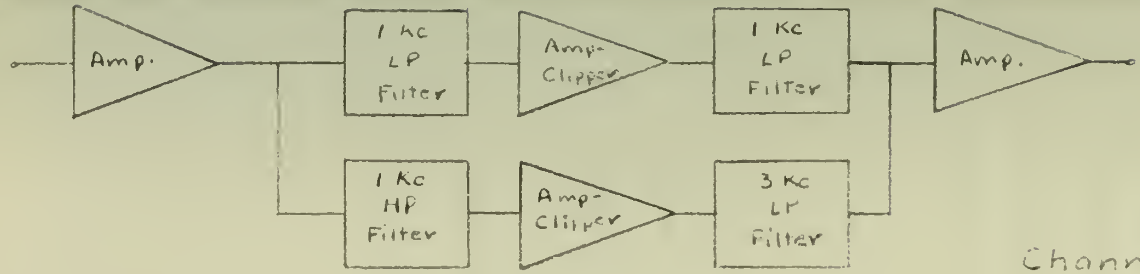
Fig. 7 Double - Integration Method

the 500 K potentiometer adjusts the overall gain, hence regulates the deviation. The output is applied directly to the grid of the phase modulator tube of the transmitter.

(b) Channel Separation Circuit

This speech clipping circuit is designed with the objectives in mind of illustrating the basic principles of the channel separation method, yet of retaining commercially practical features of a simple circuitry with a minimum of components. For this reason the octave method of channel separation is simplified to two channels, which, of course, will reduce its effectiveness.

As shown by the block diagram, Figure (8), the audio spectrum is filtered into 300-1000 cps and 1000-3000 cps channels by low-pass and high-pass filters of 12 db per octave attenuation. Each channel is then clipped and filtered by 1000 cps and 3000 cps low-pass filters respectively. As a result of clipping and filtering, harmonic distortion above the second harmonic will be small, except for third harmonics of the lower frequencies of each channel. The channels are recombined and amplified before applying to the phase modulator. Because of both phase cancellation upon recombining and the attenuation on both channels at the "cut-off" frequency, the frequency response has a pronounced null at 1000 cps. The response can be corrected to within $\pm$ 3 db in the desired frequency range by adjusting the gain of each channel by means of the cathode potentiometers in the clipper circuits. The cathode-coupled clipper has



Block Diagram
Channel Separation Method

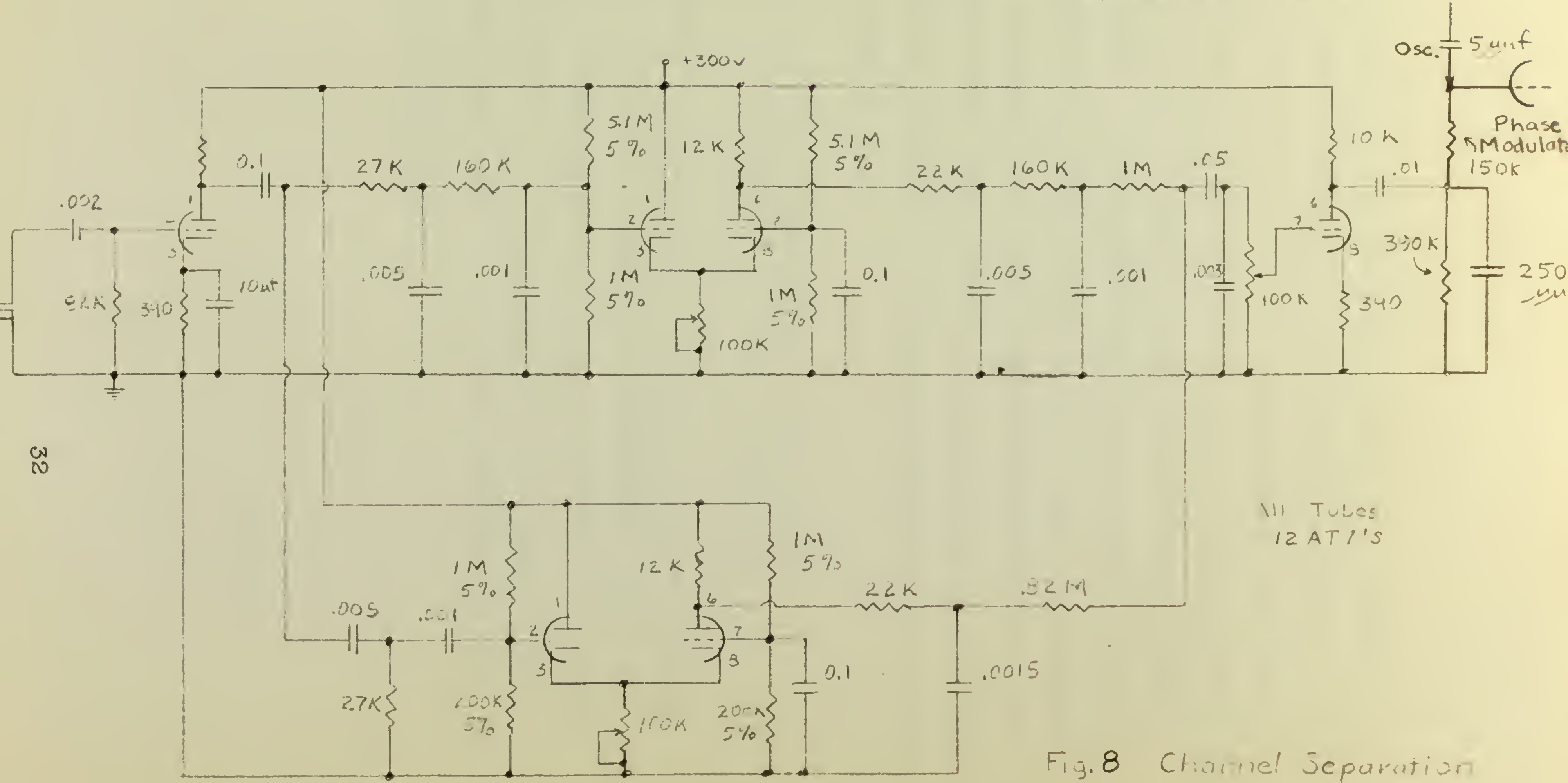


Fig. 8 Channel Separation

the advantages of voltage gain (approximately 10, as designed), excellent clipping characteristics, and high input impedance. Clipping symmetry is obtained by maintaining close tolerance in the selection of the resistors in the grid voltage dividers. Two-section RC filters are used instead of LC filters because of the advantages of cost, size and weight at audio frequencies. The time constants of each section within each filter are equal, but the value of each component is chosen to provide the sharpest "cut-off" possible by making the R_1C_2 product much smaller than that of R_1C_1 and R_2C_2. This relation can be seen by inspection of the transfer function for an integrator circuit, which is

$$\frac{1}{(1 - R_1C_1R_2C_2\omega^2) + j\omega(R_1C_1 + R_2C_2 + R_1C_2)}$$

It is important that each filter have sharp attenuation characteristics to prevent break-through on the complementary channel for excessive "overmodulation"; that is, a large signal may be clipped on one channel, yet be of such amplitude that the output of the second channel significantly increases the combined output until both channels are limiting.

CHAPTER IV

EVALUATION OF SPEECH CLIPPING CIRCUITS

1. General

In order to determine the effects of both clipping and unclipped modulation on a phase or frequency-modulated carrier, a procedure for testing and evaluating must be undertaken. Measurement of side-band amplitudes for modulation by sinusoidal, triangular, and square waves to verify experimentally the mathematical derivations described in Chapter II is of significance. Examination of the speech clipping circuits on the basis of the frequency response of an unclipped signal and a clipped complex signal are of importance for evaluating their sound quality and intelligibility. Measurements such as these give an indication of expected performance; more significant results, however, can be made by actual measurement of both the sideband amplitudes and frequency deviation of a transmitted signal which is phase-modulated by speech. Further, as a measure of speech intelligibility under various degrees of clipping, articulation tests should be undertaken. In accordance with this discussion, the following tests are described in this section:

1. Theoretical sideband measurements.
    a. sine-wave, phase modulated.
    b. triangular, phase modulated.
    c. integrated-triangular, phase modulated.

2. Frequency response.

   a. sine input, unclipped.

   b. complex input, clipped.

3. Deviation measurements.

   a. speech modulation, with input voltage of 0 db, 10 db, and 20 db relative to clipping level.

4. Bandwidth measurements.

   a. speech modulation with various degrees of clipping; as in (3).

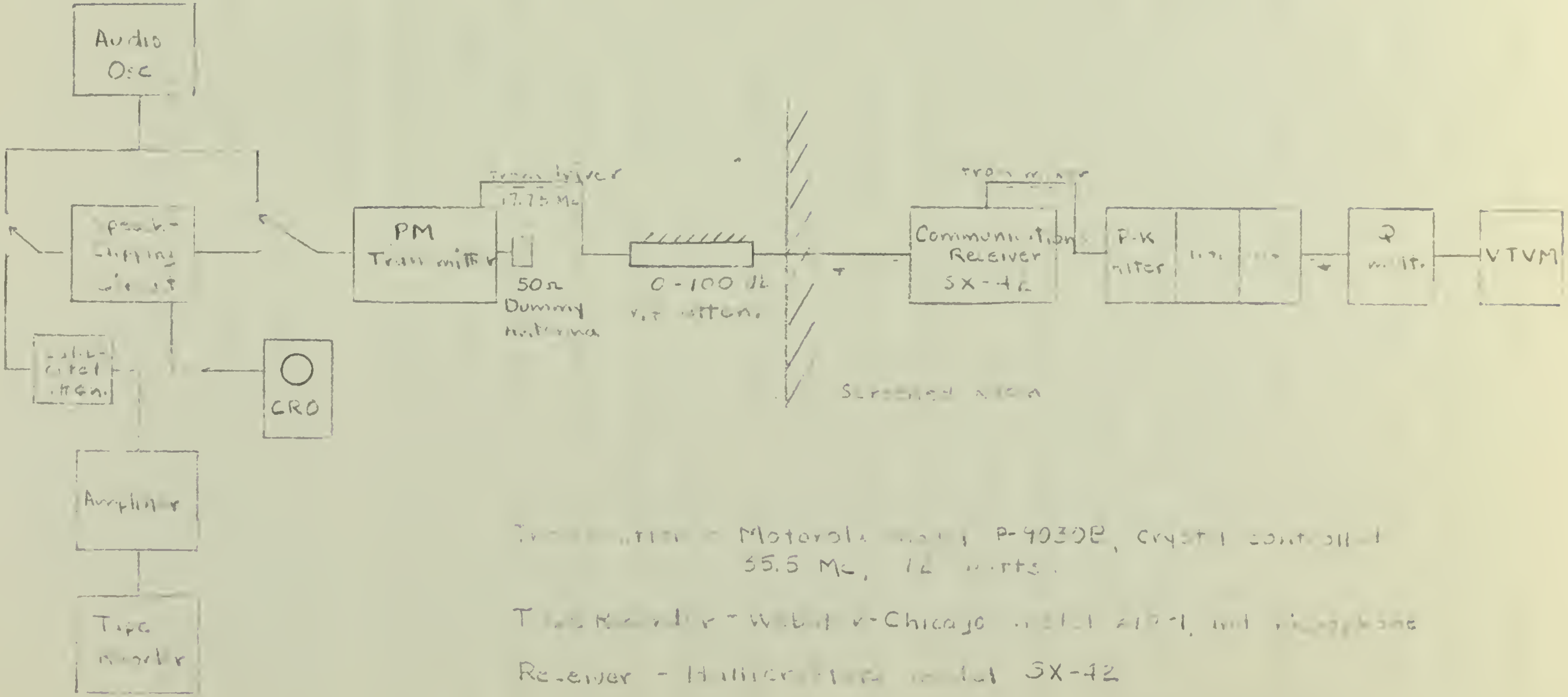
5. Articulation tests.

   a. speech modulation with various degrees of clipping; as in (3).

2. <u>Theoretical Sideband Measurements</u>

Figure (9) shows the arrangement of equipment for making this and subsequent deviation and bandwidth measurements. Important considerations for obtaining reliable measurements are: first, a high degree of nose and skirt selectivity must be available for separating adjacent sidebands and for separating small sidebands from closely spaced large amplitude sidebands; second, the range of measurements available must enable readings to at least -80 db with respect to the unmodulated carrier; and third, the accuracy of the readings must not be limited by the linearity of the receiver.

These requirements are met with the use of the receiver arrangement shown in the block diagram. The r.f. and mixer stages of an SX-42 communications receiver are used to provide a 455 kc signal



Transmitter - Motorola model P-4030B, crystal controlled
55.5 Mc, 12 watts.

Tape Recorder - Webster-Chicago model 81-1, and microphone

Receiver - Hallicrafters model SX-42

Indicating Device - HP 400C, average indicating VTVM

Fig. 9 Block diagram of measuring Equipment

for the laboratory receiver, which consists of a modified constant-K and m-derived network with 30 kc bandwidth and steep skirt selectivity followed by two i.f. stages. Use of the laboratory receiver is made on the premise that to avoid i.f. desensitization (due to overloading), the most selective circuits should precede the high-gain stages. The Q multiplier,[9] shown in Figure (10), has a nose selectivity with a Q of approximately 5000. Selectivity characteristics of the system are shown by Figure (11). With sinusoidal modulation, sidebands of 1 kc separation are easily discernible.

The operating range depends on the system's skirt selectivity, receiver desensitization characteristics, and the range of the variable r.f. attenuator. The system is designed from consideration of these factors, and as a result, reliable measurements may be made to -85 db, at which point the receiver r.f. stage overloads.

Effects of system nonlinearity are minimized by making initial readings of the carrier with 100 db attenuation in the transmission line; as the sidebands reduce in amplitude, the attenuation is reduced so as to make the indicated output constant. Neglecting linearity considerations, the indicator gain may be increased 10 db, thus making possible readings to -95 db of which the last 10 db are less reliable.

The indicator is a Hewlett-Packard 400C average-indicating vacuum tube voltmeter. The receiver and indicator equipment are placed in a screened room, in order to minimize reception of undesired

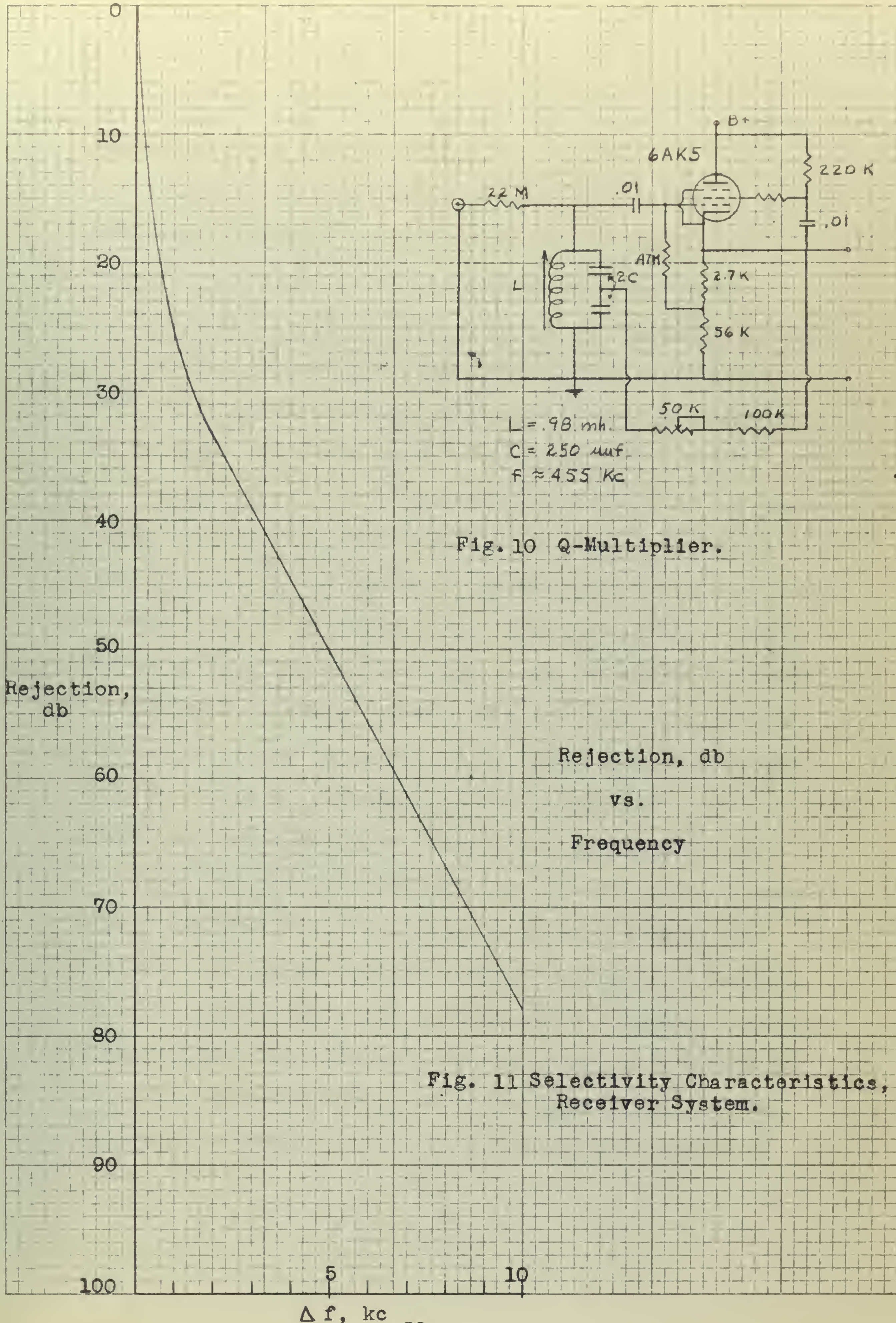


Fig. 10 Q-Multiplier.

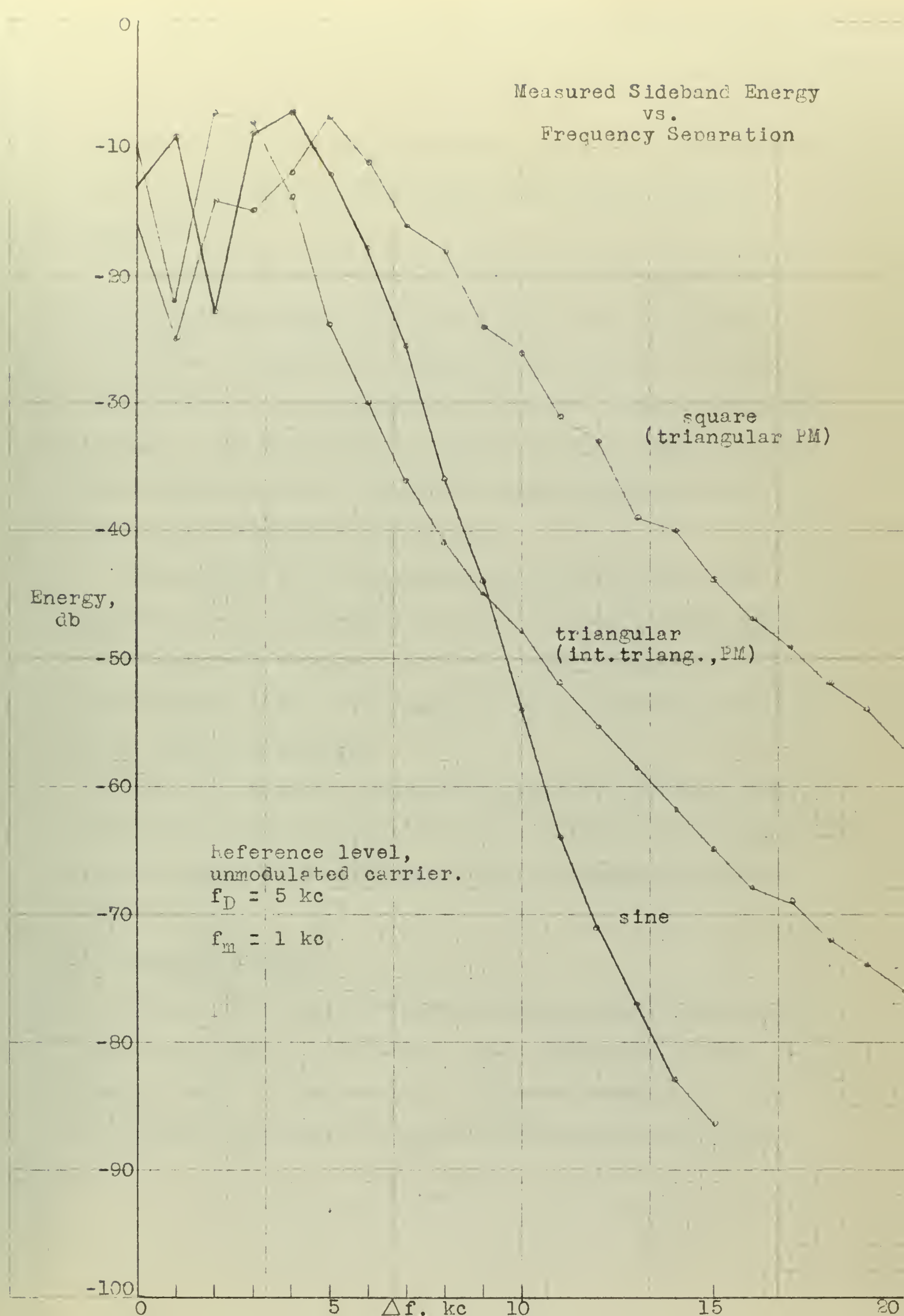
Fig. 11 Selectivity Characteristics, Receiver System.

radiation.

Sine wave modulation is obtained by use of the indicated audio oscillator. Triangular and integrated-triangular wave modulating voltages are obtained by driving a modified IDC unit beyond the clipping point. The IDC is modified to make the triangular waveform more linear by increasing the time constant of the integrator circuit; the integrated-triangular waveform is obtained by integrating the IDC output with a 200 cps RC network.

Measurements are made with a modulating frequency of 1000 cps, deviation of 5 kc, and modulation index of 5. The deviation for sine wave modulation is adjusted by increasing the modulating voltage so that the carrier is reduced to the value corresponding to $J_0(5)$. The deviation for the other waveforms is adjusted to 5 kc by making the modulating voltage for the triangular wave 1.57 x the sine wave amplitude of the same deviation, and the integrated-triangular wave .78 x the sine wave. A check may be made by adjusting each of the latter waveforms to have equal maximum slopes on an oscilloscope.

Figure (12) shows a comparison of the envelopes of the measured sideband amplitudes with the theoretical envelopes of Figure (5). Very close agreement is noted for sinusoidal modulation to the tenth sideband, where the discrepancy is 4 db, with a gradually increasing error for higher order sidebands. This difference is attributed to the selectivity characteristics of the measuring system, which has, for example, a rejection of 55 db at 6 kc separation, whereas the



Measured Sideband Energy
vs.
Frequency Separation
square
(triangular PM)
triangular
(int.triang.,PM)
sine
Reference level,
unmodulated carrier.
f_D = 5 kc
f_m = 1 kc
Energy,
db
0
-10
-20
-30
-40
-50
-60
-70
-80
-90
-100
0
5
10
15
20
Δf, kc

difference between the fourth and tenth sidebands is theoretically 49 db. Similar errors for other modulating waveforms will be negligible.

Comparison of the triangular phase-modulated envelope with that of a square wave frequency-modulated carrier shows agreement to -2 db at 12 kc separation. The error, which increases for higher order sidebands, is attributable to a nonlinear triangular modulating voltage. A null was observed at each odd harmonic except the fifth; for a modulation index of precisely 5, these odd harmonics will theoretically reduce to zero amplitude.

The spectrum for the integrated-triangular wave shows less agreement with its FM counterpart than do the previous comparisons. Nevertheless, the envelopes are of very similar shape and in fact, nearly parallel. Such a discrepancy is believed due to nonlinearity of the modulating waveform.

Thus it is seen that the observed spectra are in close accord with the theoretical spectra and, within the bounds of experimental accuracy, verify the derivations described in Chapter II, for the examples chosen.

3. <u>Frequency Response</u>

Figure (13) shows the measured frequency response of the speech amplifier circuits for a sinusoidal input voltage that is below the clipping level. In comparison, the IDC has a response flat within - 3 db from 250-3750 cps; the response of the double integrator is

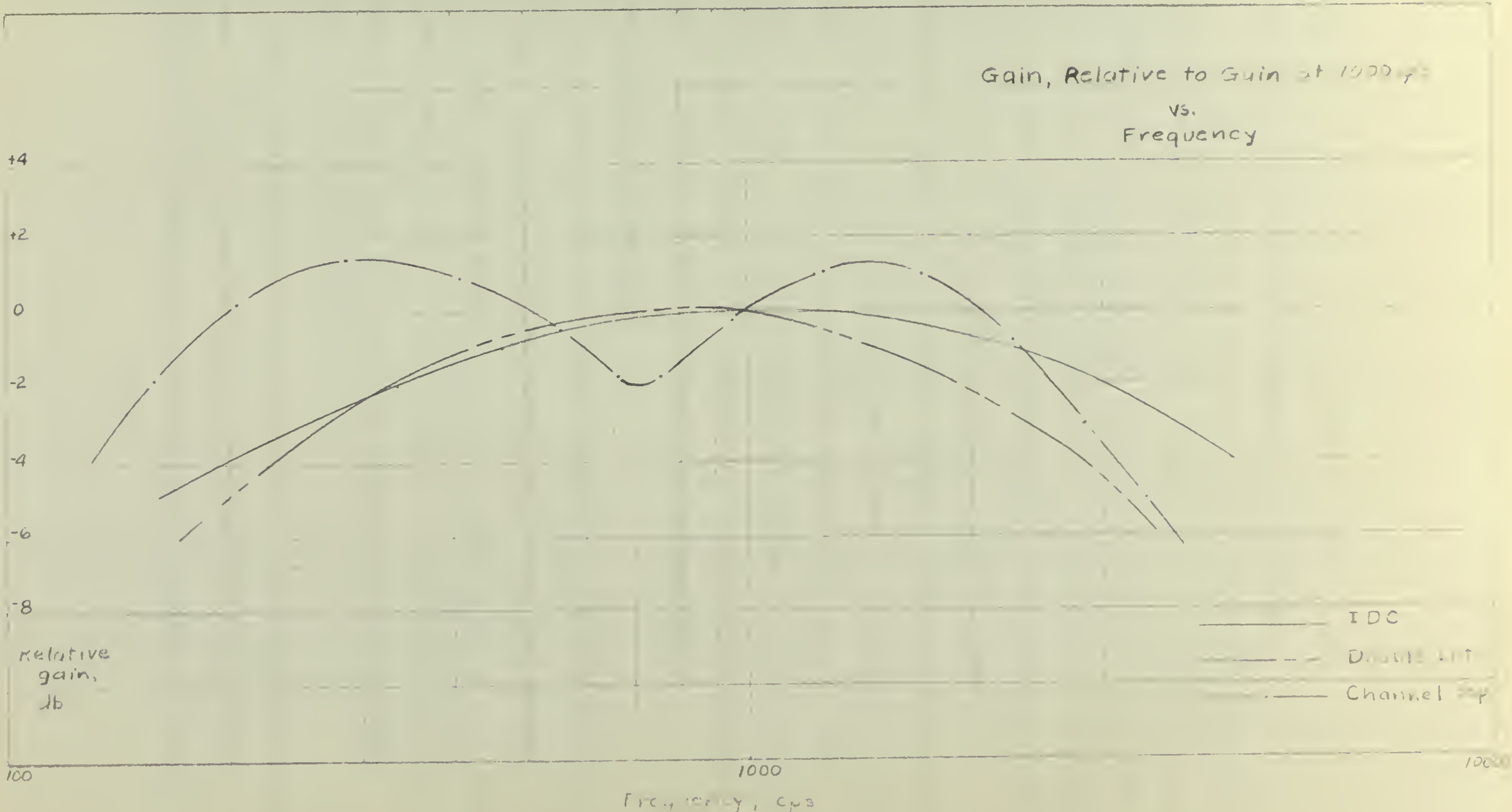


Fig. 13 Frequency Response, Unclipped Sine Wave

270-2500 cps, and the channel separation method shows a response from 180-2700 cps, with a pronounced dip at 700 cps.

Analysis of clipped modulation is made with a complex waveform. The response of each system with an input of two sinusoidal voltages of frequencies 350 cps and 2000 cps in the amplitude ratios of 2:1, 1:2, and 10:1 is plotted in Figure (14). Significance may be attached to the relative suppression characteristics of each clipper circuit. Ideally the ratio of the output voltages should be identical to the input ratios. The suppression depends on the ratio of the voltages at the input to the clipping stages; because of the emphasis preceding the clipping stages, the suppression characteristics will differ from the ideal. In comparing the three systems it is apparent that the clipping response of the channel separation and IDC circuits show a marked improvement over the double integrator.

4. Deviation Measurements

A communications-quality tape recorder is used with the speech-clipping circuits to modulate the PM transmitter. The recording is a short, repeated paragraph spoken by a male voice at approximately constant peak amplitudes. The deviation is measured on an oscilloscope, which is used to indicate the output from the calibrated discriminator of the receiver.

The gain of the recorder is adjusted so that the speech waveform is at the point of peak clipping by each speech-clipping circuit. The calibrated attenuator is then adjusted to increase the speech

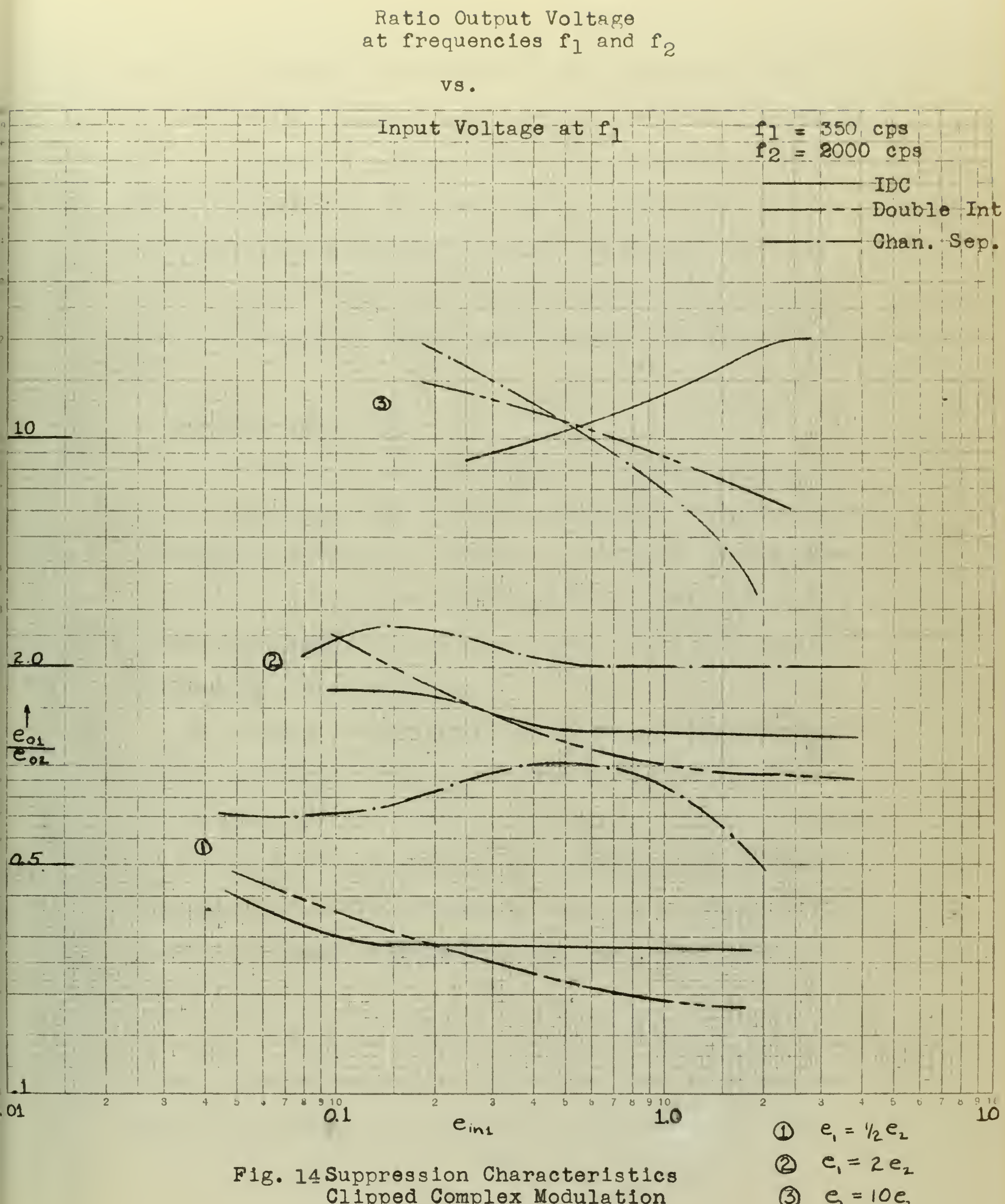


Fig. 14 Suppression Characteristics
Clipped Complex Modulation

level to 10 db and 20 db relative to the clipping level, thus permitting measurements at various practical degrees of "overmodulation". A comparison of the deviation limiting characteristics is shown by Figure (15). It is seen that all three methods prove effective in controlling the deviation. The performance of the channel method is slightly preferable at higher modulation levels; characteristics of the double-integrator circuit prove least desirable.

5. Bandwidth Measurements

The procedure for making bandwidth measurements is identical to that for the preceding deviation measurements. The peak deviation for unclipped modulation is adjusted to be 5 kc; as the input power is increased in 10 db steps, the deviation increases as shown in the preceding paragraph. Equipment arrangement, shown in Figure (9), is described in paragraph (IV-2).

The recorded sideband amplitudes are the peak readings of an average-indicating voltmeter and plotted in db referred to the unmodulated carrier. The voltmeter indication varies 5 to 10 db with the modulation peaks on all sideband frequencies; thus, neglecting the meter inertia and time constant, the recorded readings are the peaks of the average sideband amplitudes converted to power in decibels.

Figure (16) shows the bandwidth characteristics of the IDC. Of particular interest are the curves at 0 db and 20 db with respect to the clipping level, adjusted for equal deviation. The

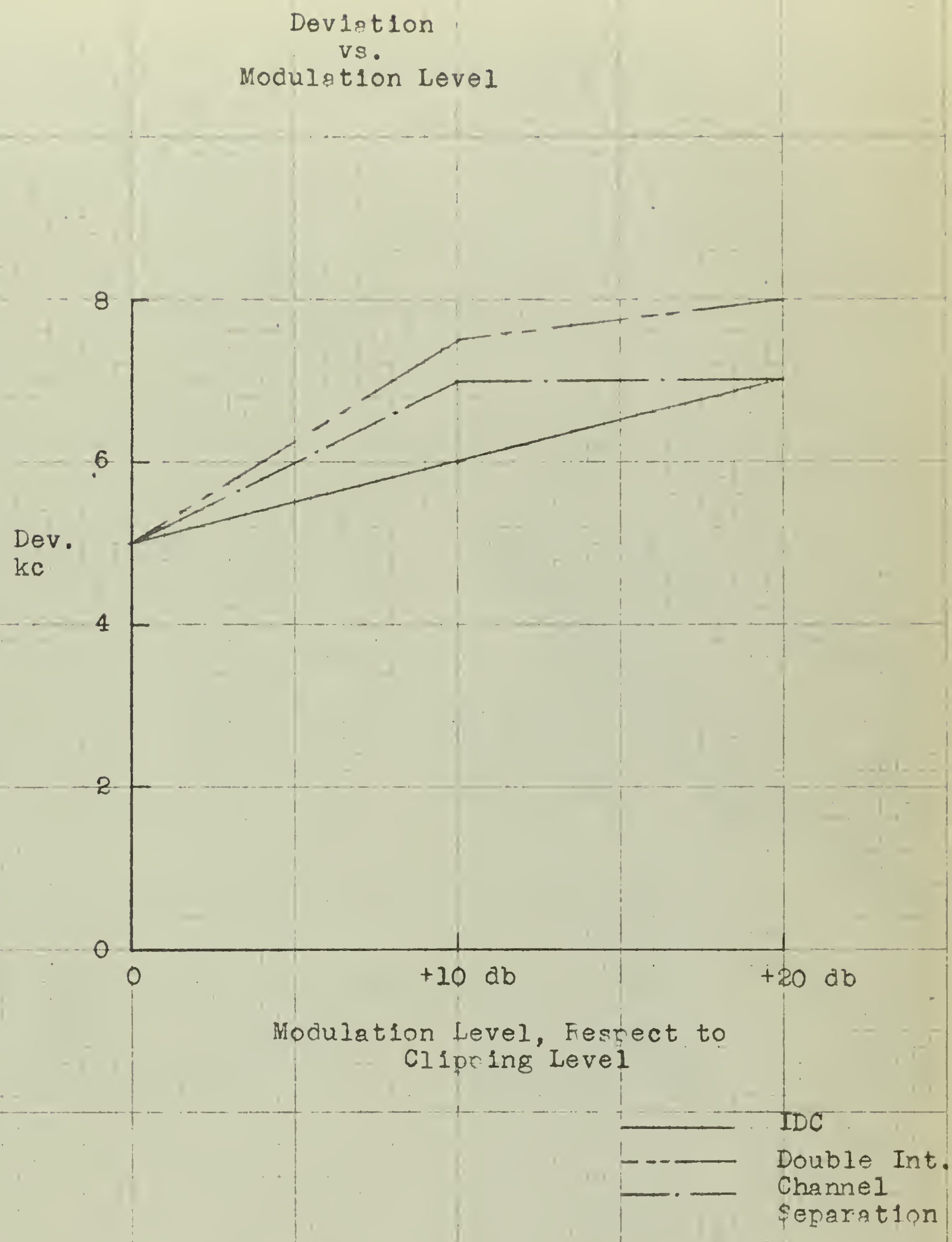


Fig.15 Deviation Characteristics

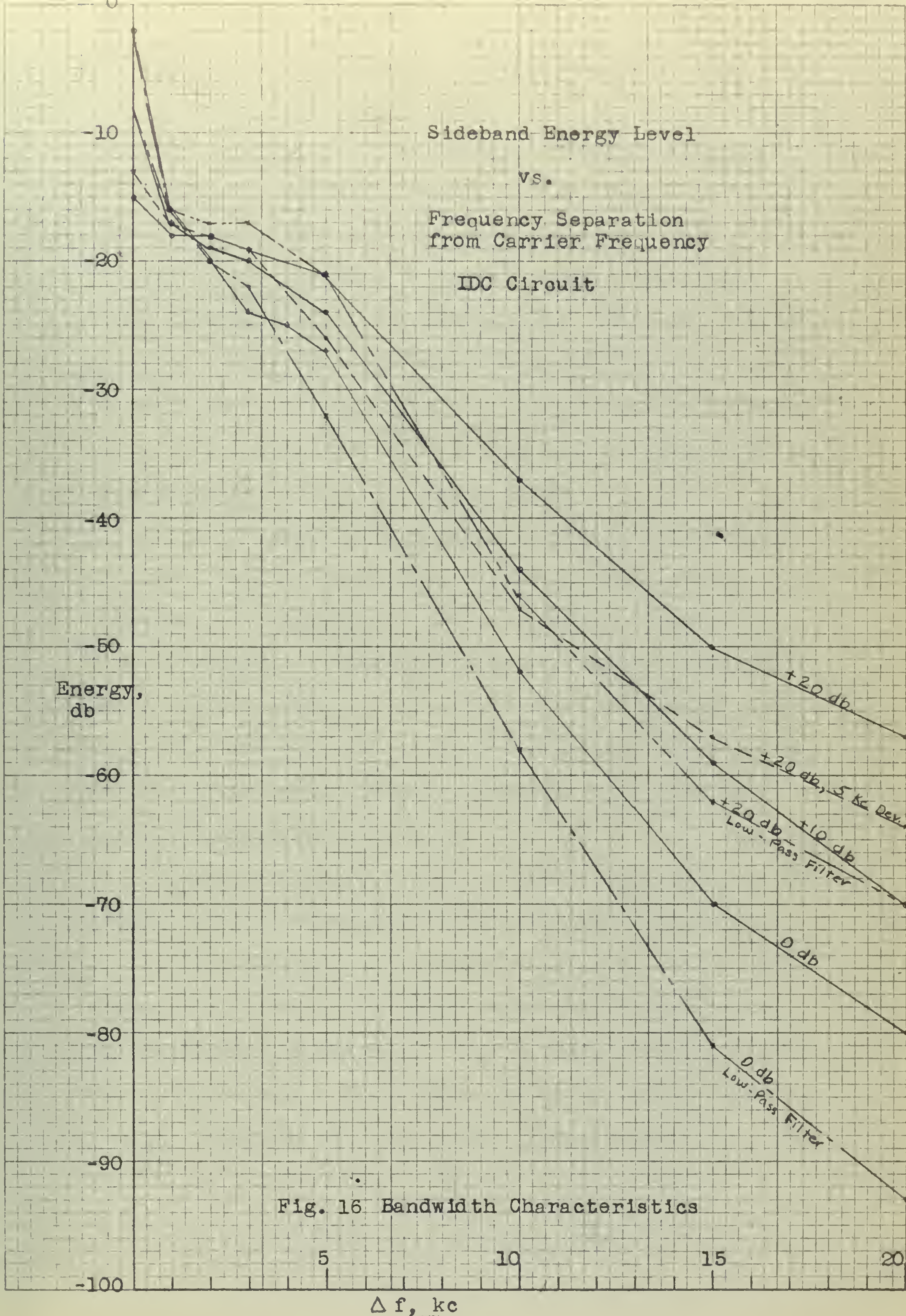


Fig. 16 Bandwidth Characteristics

difference in levels of the two curves at a frequency separation of 20 kc from the unmodulated carrier is 16 db, showing the deleterious effect of clipped modulation on the bandwidth.

The effect of increased deviation is shown by comparing the above 20 db curve of 5 kc deviation with the 20 db curve obtained as stated in the first paragraph of this article (7 kc deviation). The energy level at 20 kc separation is 6 db greater for the latter curve. A 12 db improvement at 20 kc separation is obtained with the addition of a simple 3 kc low-pass filter.

Figure (17) shows the bandwidth characteristics of the double-integrator clipping circuit. Despite its increased deviation, this method has a 6 db improvement over the IDC for the 20 db curves, at 20 kc separation. Clipped modulation with constant deviation is seen to increase sideband energy content only 6 db. The IDC, with the addition of a 3 kc filter, nevertheless measures a 6 db improvement over this clipping circuit, attributable to the superior deviation characteristics of the IDC.

The channel separation circuit, as seen by Figure (18), reveals a marked bandwidth improvement over the other methods. The level at 20 kc separation is 22 db less than with the IDC. This is attributable to its inherent reduction of higher order clipping products, its good deviation characteristic, and its narrow frequency response. The IDC with a low-pass filter shows very similar characteristics on unclipped modulation, and 10 db greater sideband energy at 20 kc

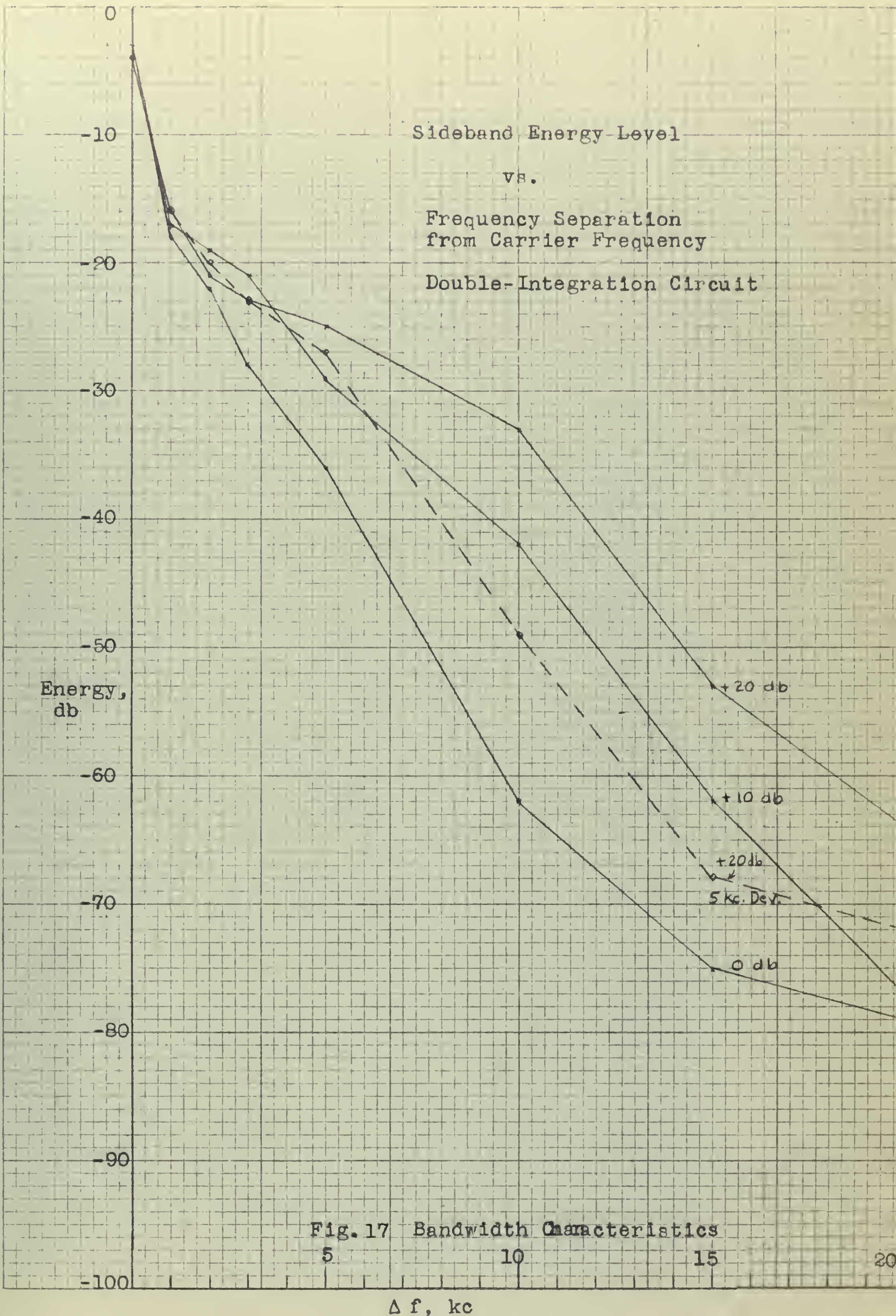


Fig. 17 Bandwidth Characteristics

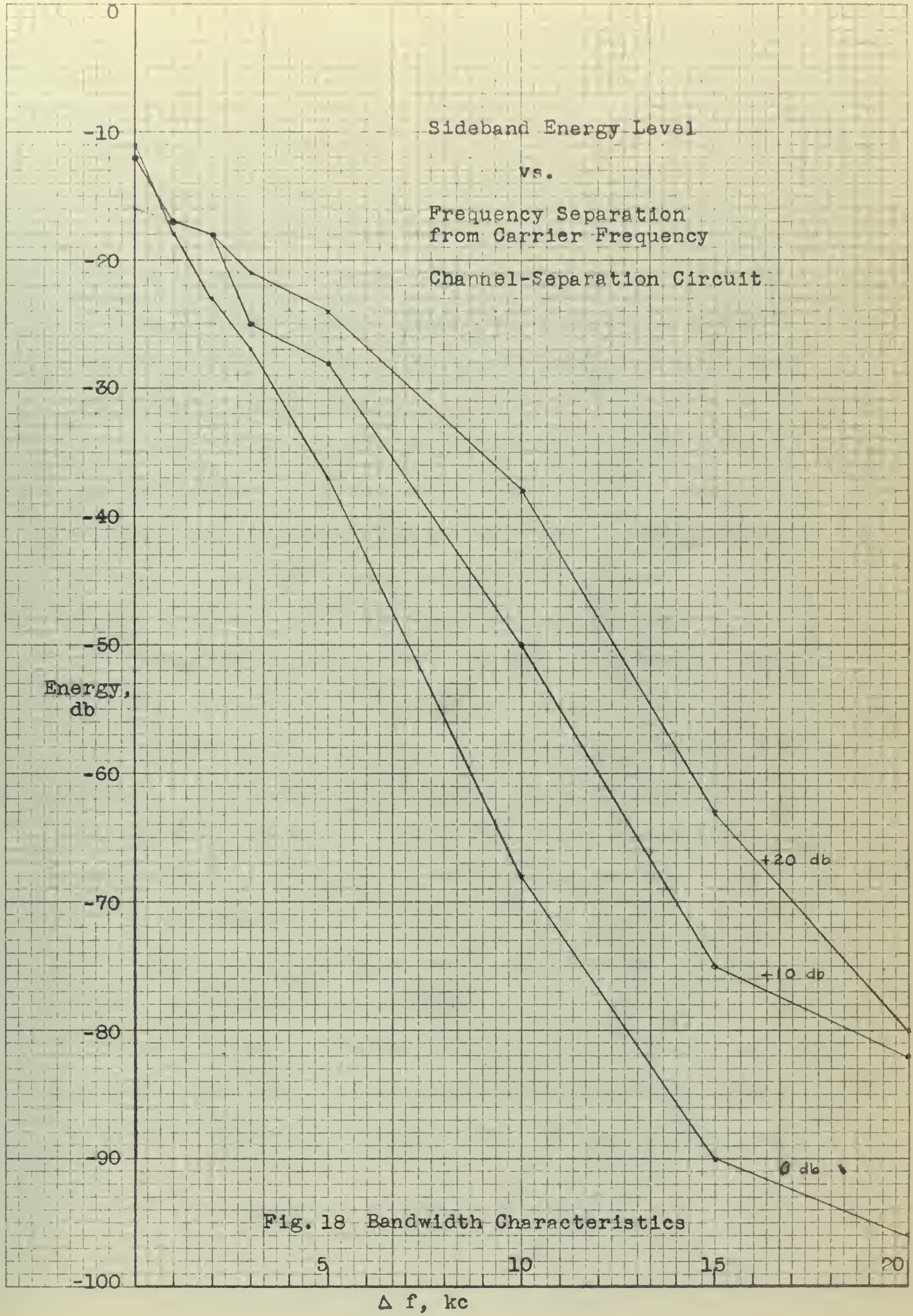


Fig. 18 Bandwidth Characteristics

separation with 20 clipping.

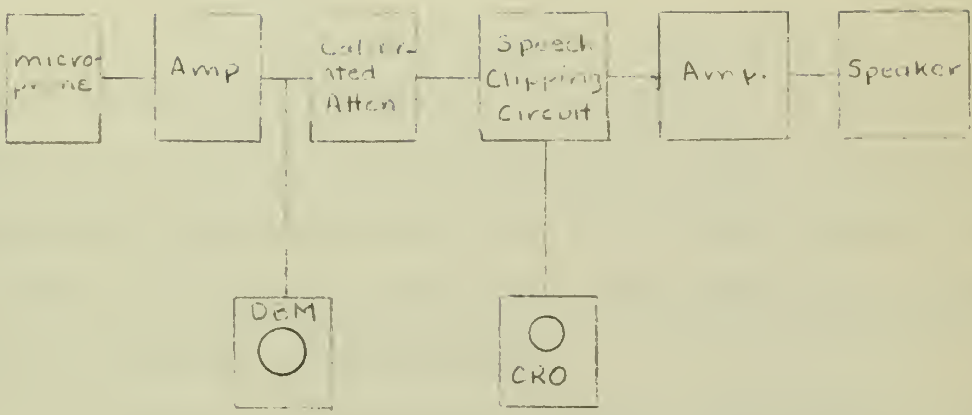
6. Articulation Tests

Syllable articulation is compared for the three circuits with 0db, 10 db, and 20 db levels with respect to the level at which clipping of the peaks begins. Methods of testing are in accordance with the standard articulation lists as devised by Bell Laboratories[10] for testing telephone systems.

A test syllable consists of three fundamental sounds in the order of consonant-vowel-consonant. Each syllable is spoken with a short introductory sentence to resemble actual connected speech. Twenty-two consonants are used for the initial and final sounds, and the vowels are listed with twenty-two characteristic sounds, thus forming twenty-two syllables. The test is conducted by writing each sound on a card and placing the cards in three boxes. The cards are then selected at random. This procedure is repeated three times to obtain a list of sixty-six syllables. Discrete sentence intelligibility may then be obtained from the per cent syllable articulation by a curve shown in the referenced article.

The arrangement of equipment for the tests is shown in Figure (19). In order to reduce the effects of room acoustics and external noise, personnel are located outside.

One caller and one observer conducted the tests after a training period of three days. It is recognized that a training period of at least two weeks duration would be more desirable. Periodic tests



Microphone--(1) Crystal, model 32-401 Shure
(2) Carbon, model CB12E Shure

Loudspeaker--10" (Hallicrafters, SX-42)

Fig. 19 Articulation Testing Equipment

of the articulation in air were made, however, to account for the effects of personnel improvement. It would also be advantageous to use two callers and at least two observers to obtain additional statistical information.

Results of the tests indicate no difference between the three circuits for the clipping levels chosen. Using the crystal microphone, sentence intelligibility is extrapolated to be approximately 99% for each clipping level with each system. The only detectable difference is the notably bassy sound of the double integrator when clipping. Although all systems sound poorly under 20 db clipping, the test syllables are understandable.

Of interest is the comparison between a medium quality crystal microphone and a carbon microphone of the type used in mobile equipment. Although 91% syllable articulation is recorded for the carbon microphone at 0 db relative to the clipping level, only 73% is recorded under 20 db clipping. In contrast, no difference is noted between 0 db and 20 db clipping levels with the crystal microphone. This indicates that the microphone may contribute appreciably to the loss of intelligibility when the speech wave is being clipped.

CHAPTER V

CONCLUSION

The investigation may be summarized as follows:

1. The bandwidth of a frequency-modulated signal is a function of

   a. deviation

   b. repetition frequency of modulating voltage

   c. waveshape of the modulating voltage.

2. In order to restrict the transmitted bandwidth, the deviation must be limited.

3. If limiting the deviation results in frequency modulating with a square wave, the bandwidth of significant sideband amplitudes is greatly increased over that produced by modulating with a sine wave of the same deviation. It is concluded that the high harmonic content of the square wave, which determines both its maximum slope and abrupt corners, is responsible for its increased bandwidth, and that the significant bandwidth may be reduced by attenuating the higher frequency components. This may be extended to any irregularly shaped modulating waveform.

4. Two general methods of deviation limiting are: (1) speech compression, and (2) speech clipping. Speech clipping is considered more desirable from the viewpoints of its effective deviation limiting, instantaneous action, and inherent circuit simplicity. In order to reduce the high frequency harmonic content produced by clipping and hence reduce sideband splatter, a low-pass filter is desirable.

Further, low-pass filtering the distorted speech waveform improves the intelligibility of the speech.

5. Consideration of the interference effects of modulation by clipped speech indicates that significant peak noise powers extend into the neighboring channel, but the total energy in the adjacent channel is nevertheless small.

6. When a complex modulating waveform is operated on by a nonlinear circuit, such as a clipper, analysis by superposition of single frequency sine waves is invalid, since the ratio of the amplitudes of the component frequencies before clipping is not equal to the ratio of the same frequencies after clipping. Suppression of the smaller amplitude components makes speech pre-emphasis desirable prior to clipping.

7. Three speech clipping principles are tested. It is determined by measurement upon a carrier modulated by speech that the side frequency amplitudes produced by the audio channel separation technique decrease more rapidly than with the IDC or double integration methods; sideband splatter produced by the IDC is significantly decreased with the addition of a simple R-C 3000 cps low-pass filter.

8. Articulation tests show approximately 99% sentence intelligibility for each of the three clipping circuits under clipping levels with peak input power as much as 20 db greater than the level at which clipping was first observed. Sound quality is considered

objectionable at the 20 db clipping level.

9. With the addition of a low-pass filter as described above, the IDC performs approximately as well or better than the other circuits developed, and because of its inherent circuit simplicity proves to be the most economically feasible. It is conceivable that the channel separation method, if carried out to its full capabilities by addition of at least two channels and with sharp filter characteristics of -18 db/octave attenuation, would prove to be a relatively high quality system with good deviation characteristics and fidelity.

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Fig 12

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G257 Geary 20741
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